

Houyuan VoIP PBX User Manual



HOUYUAN® IPPBX Product Guide

Version: 1.0

2012-09-10

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Contact Houyuan

The Introduction of Houyuan

Founded in 2009, Houyuan technology has been always endeavoring in the R&D and manufacturing of the internet communication terminals. The product line of Houyuan includes IP Phone, IP PBX, POE Switches, Wireless Router, Asterisk Card...

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Tel	+(86)755-66802184
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Chapter 1 the Introduction of IP0x

Overview of the IP0x

The IP0x is a complete Asterisk Appliance with one dual port FXO or FXS module. It is an embedded open source Linux system with built-in SIP/IAX2 proxy server and NAT features. It provides a solid, uniform platform for traditional PSTN communications as well as VoIP communications.

Targeting for SOHO user and SMB market with an easy to use graphical interface, IP0x provides a cost-saving solution on their telecommunication/data needs. With IP0x, company with branch offices in different countries can be easily combined together to work like a virtual single office through Internet.

Features

- Open Source Asterisk IP PBX
- High performance OSLEC (Open Source Line Echo Canceller)
- Configurable IVR menu
- Voice Mail, Voicemail to Email
- Call forward, Call waiting, Call transfer
- Call conference
- Call queues, Ring group
- SIP trunk, IAX trunk, PSTN analog trunk
- Call Detail Record
- Access via: SSH/telnet/web
- Firmware upgradable via web page
- 50+ available SIP/IAX2 extensions
- 20 concurrent calls

Applications

- SOHO/SMB telephony system
- Hosted service
- FAX terminal
- IVR system

Interface

- 1&2 *RJ45 port
- 1 * Power port
- 2 -8* RJ11 port (FXS/FXO interchangeable)
- 1-4 * Dual port FXO/FXS module slot

Hardware

- CPU: 400MHz Blackfin 532 Chip
- 1-8 analog (FXO/FXS) module interface

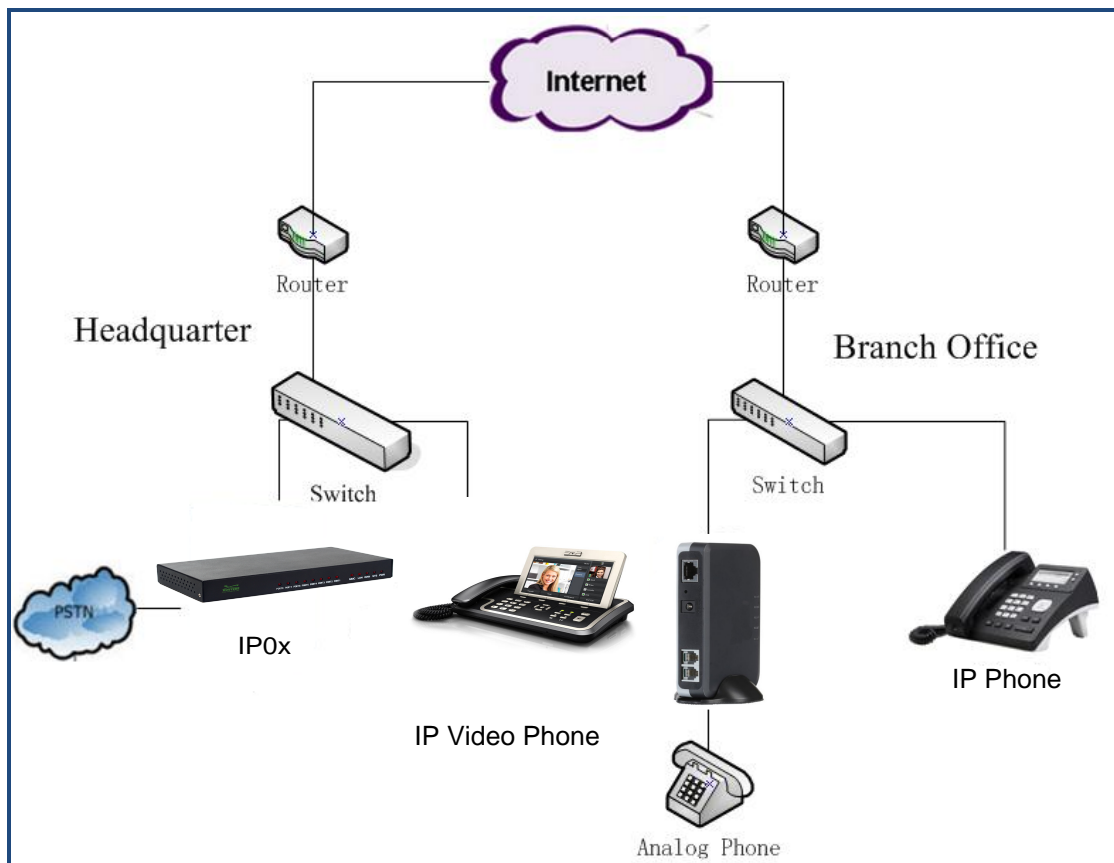
NAND flash 256 MB

SDRAM 64MB

System

Open Source uClinux

For the usage of IP0x in VoIP field, you can refer to the following network topology.



Chapter 2 Access to the IP0x

You need a PC to access to the IP0x, there are four ways for you to access the IP0x:

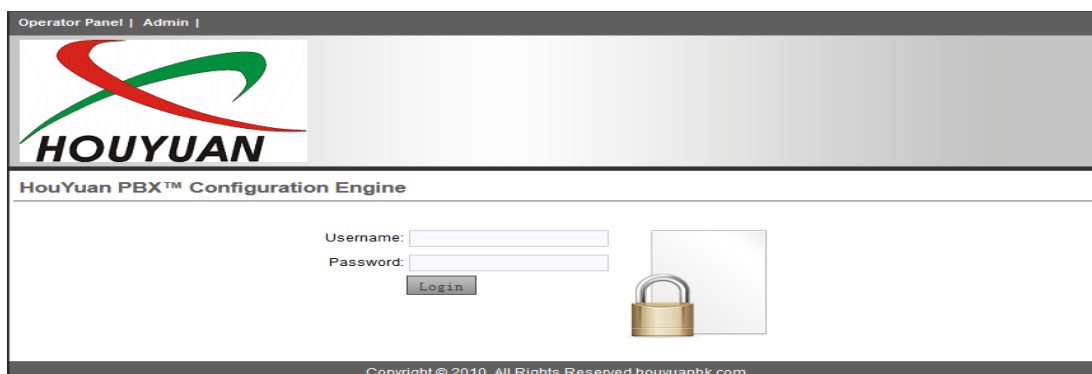
1. Web page access by browser
2. SSH access by putty
3. Access by browser with Fallback IP Address
4. Console port access by RS232 console cable

In order to access to IP0x by the first three ways, you have to check that if your network connection between IP0x and PC is OK. If you do not have network connection between IP0x and PC, you can try to use the last way to access to IP0x and change the IP address for IP0x.

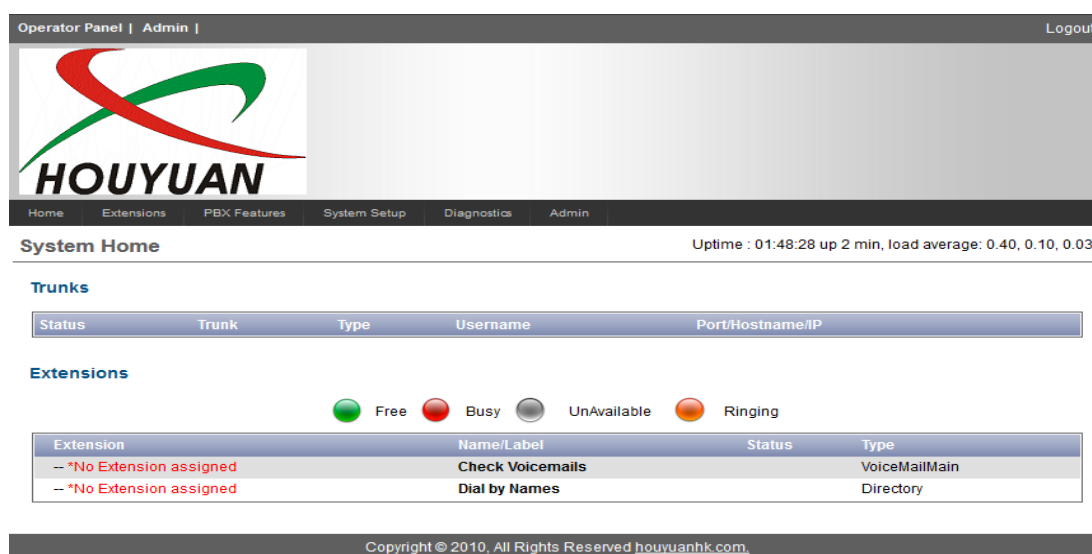
2.1 Web Page Access by Browser

It is the most convenient and common way to access the IP0x, you just need to open your browser and input the IP address of IP0x WAN port (the default IP address is 192.168.1.100). You would better use Firefox instead of IE, because there are compatible issues.

Then input the default Username: admin; Password: admin in the presented screen like the following:



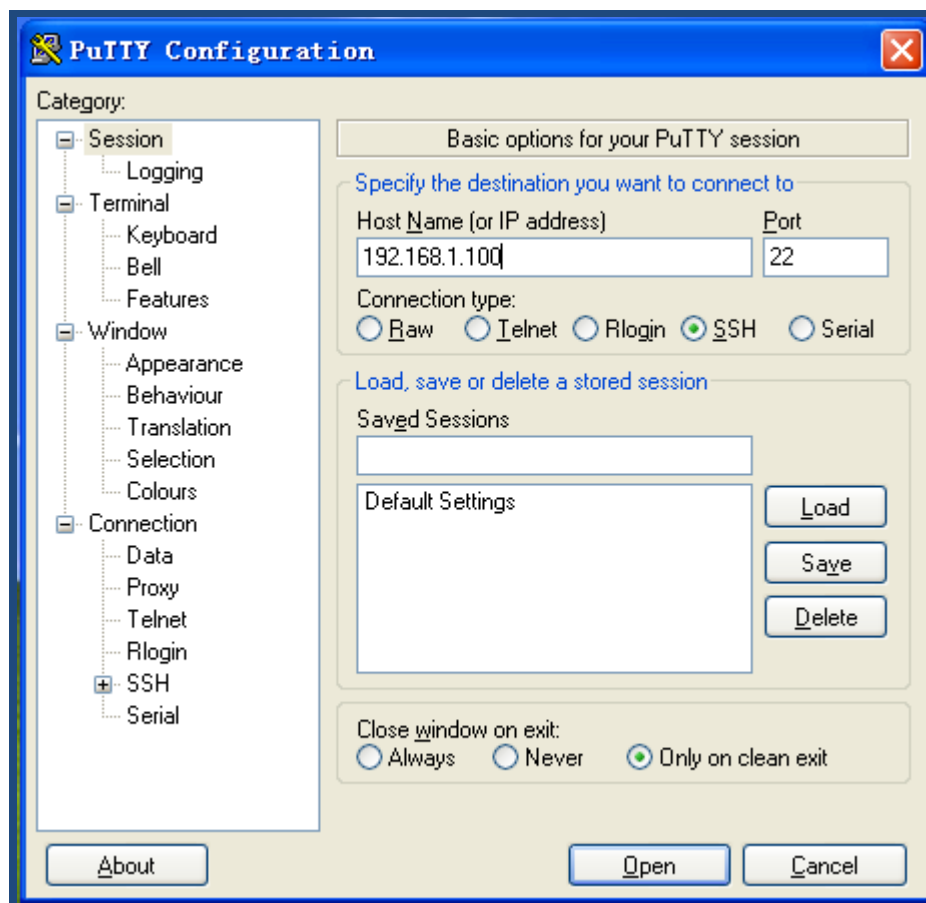
When you login successfully, you can get the configuration web page as below:



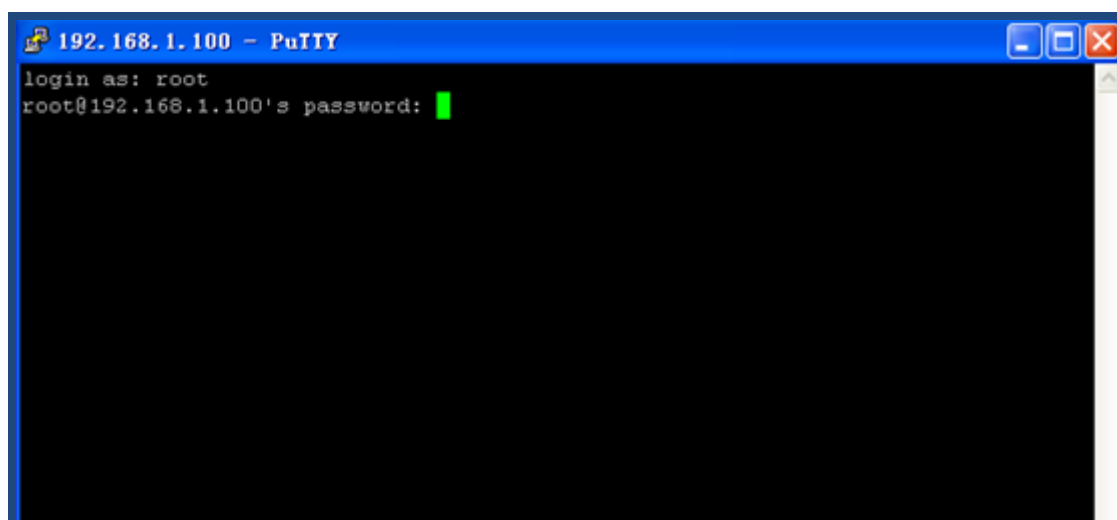
2.2 SSH Access by Putty

Logging into IP0x by SSH, you can configure IP0x by Linux command.

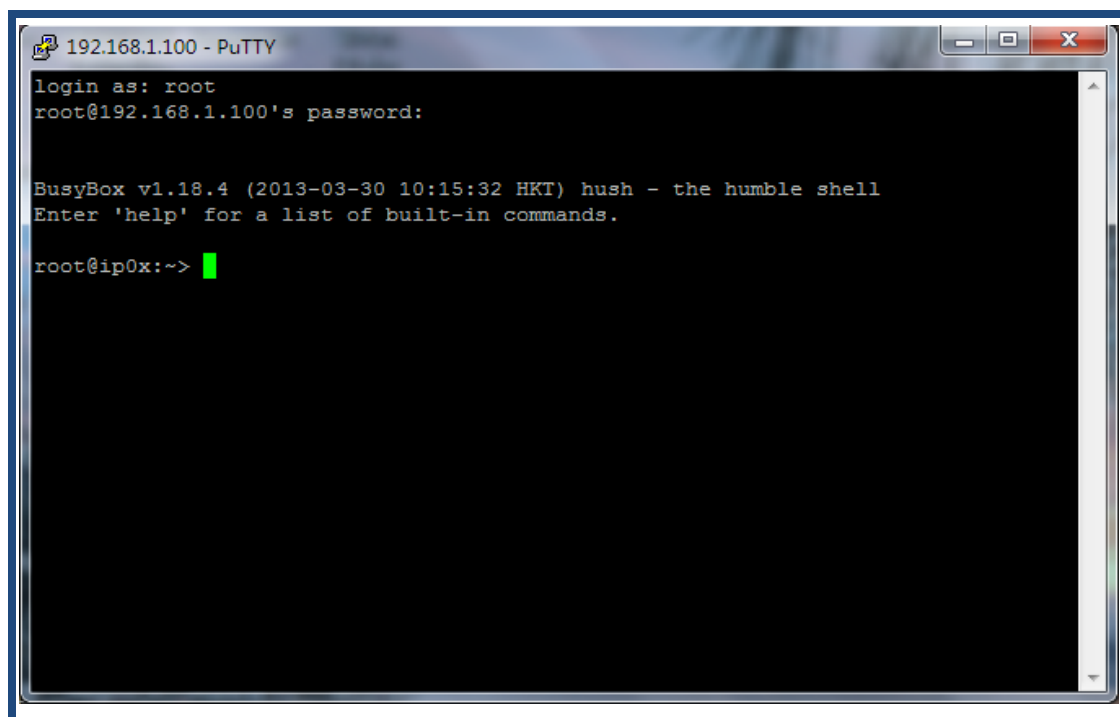
- 1) Please open your putty software, and input the IP0x IP address in the **Host Name** textbox, input port number in the **Port** textbox, click the **SSH** Connection type, then click **open** button. Please refer to the following screen:



- 2) Please input username: root, and the default password: uClinux in the following screen, you can access to IP0x successfully.



When you log into IP0x successfully, you can get the following illustration:



2.3 Access by Browser with Fallback IP Address

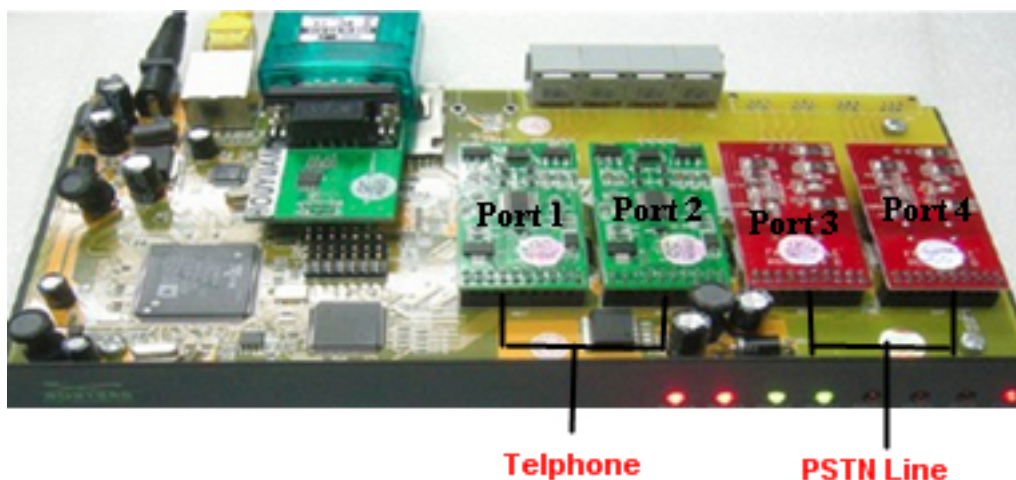
. If you forget

the IP Address of IP0x you have set up, you can use the fallback IP Address: 172.31.255.254/30. Before logging into IP0x, please set up the IP Address of your PC: 172.31.255.253 and SubMask: 255.255.255.252. At last, you can open your browser and enter:172.31.255.254 to log into the web page of IP0x.

2.4 Console Port Access to IP0x

If you do not have network connection between IP0x and PC, you can try to access to IP0x by console port. Please try to do as the following steps:

1. Please connect the console port of IP0x to your PC's console port with RS232 console cable, you can refer to the following illustration:



2. Please run your Hyper Terminal, and set up the console port like the following:

Bits per second: 115200

Data bits : 8

Parity: None

Stop bits: 1

Flow control: None

3. Change the IP Address by Hyper Terminal

The default IP address of IP0x is 192.168.1.100. Your network may have a different IP address range such as 192.168.10.xx. In this situation, you can not access to IP0x by putty and browser if you do not change the IP0x IP address. So you have to change the IP address for IP0x by Hyper Terminal to make it in the same network segment as your LAN.

After you have accessed to IP0x by Hyper Terminal, please use the following command to change the IP address for IP0x.

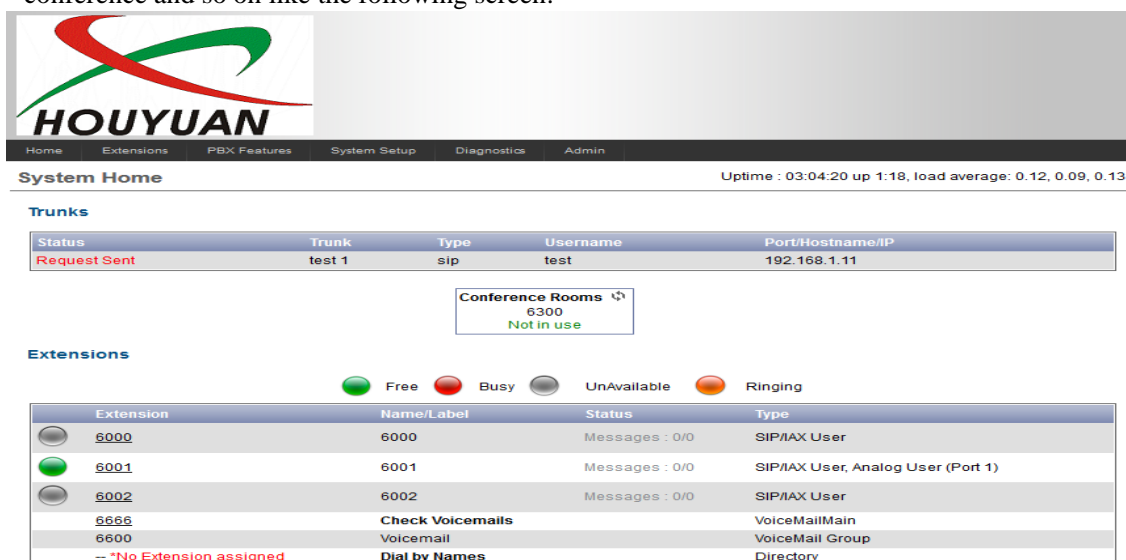
```
root:~> ifconfig eth0 192.168.1.151(the IP address you want to set for IP0x)
```

By this way, the IP address you set for IP0x is temporary, it will recover to the original default IP address after rebooting. If you want to give a static and permanent IP address for IP0x, you can try to set it in web GUI, for detail steps please refer to chapter 3.

Chapter 3 Configure IP0x by Web GUI

3.1 System Status

In the system status screen, it displays the functions you configured, such as: trunks, extensions, conference and so on like the following screen:



The screenshot shows the 'System Home' page of the Houyuan Web GUI. The page includes a navigation bar with links: Home, Extensions, PBX Features, System Setup, Diagnostics, and Admin. The 'System Home' section displays system uptime and load average. Below this, there are sections for 'Trunks' and 'Extensions'.

Trunks

Status	Trunk	Type	Username	Port/Hostname/IP
Request Sent	test 1	sip	test	192.168.1.11

Conference Rooms

6300	Not in use
------	------------

Extensions

Legend: Free (Green), Busy (Red), UnAvailable (Grey), Ringing (Orange)

Extension	Name/Label	Status	Type
6000	6000	Messages : 0/0	SIP/IAX User
6001	6001	Messages : 0/0	SIP/IAX User, Analog User (Port 1)
6002	6002	Messages : 0/0	SIP/IAX User
6666	Check Voicemails		VoiceMailMain
6600	Voicemail		VoiceMail Group
-- *No Extension assigned	Dial by Names		Directory

3.2 Configure Hardware

In the configure hardware page, it includes the following components: analog hardware, tone region, advanced settings.

Analog Hardware

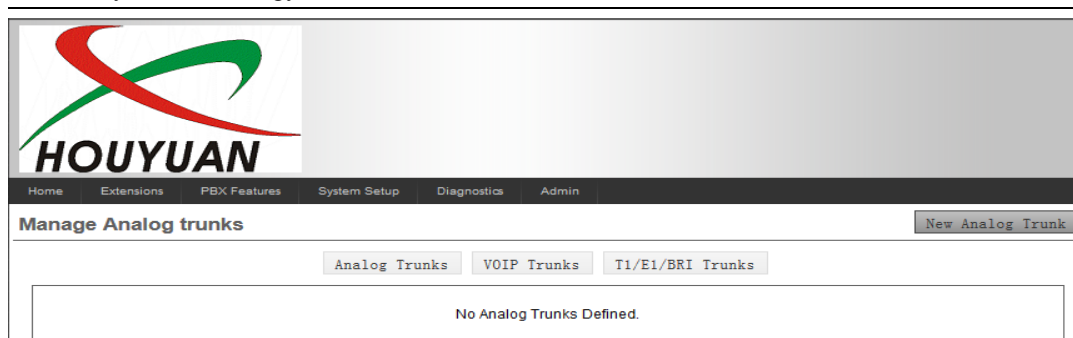
When you boot the IP0x, which will detect the FXO and FXS modules automatically, the analog hardware component displays the modules which are detected correctly.

Tone Region

You should select the tone region according to your country, if it does not have your country's name in the dropdown list, please ask your service operator which kind of tone region is used in your area.

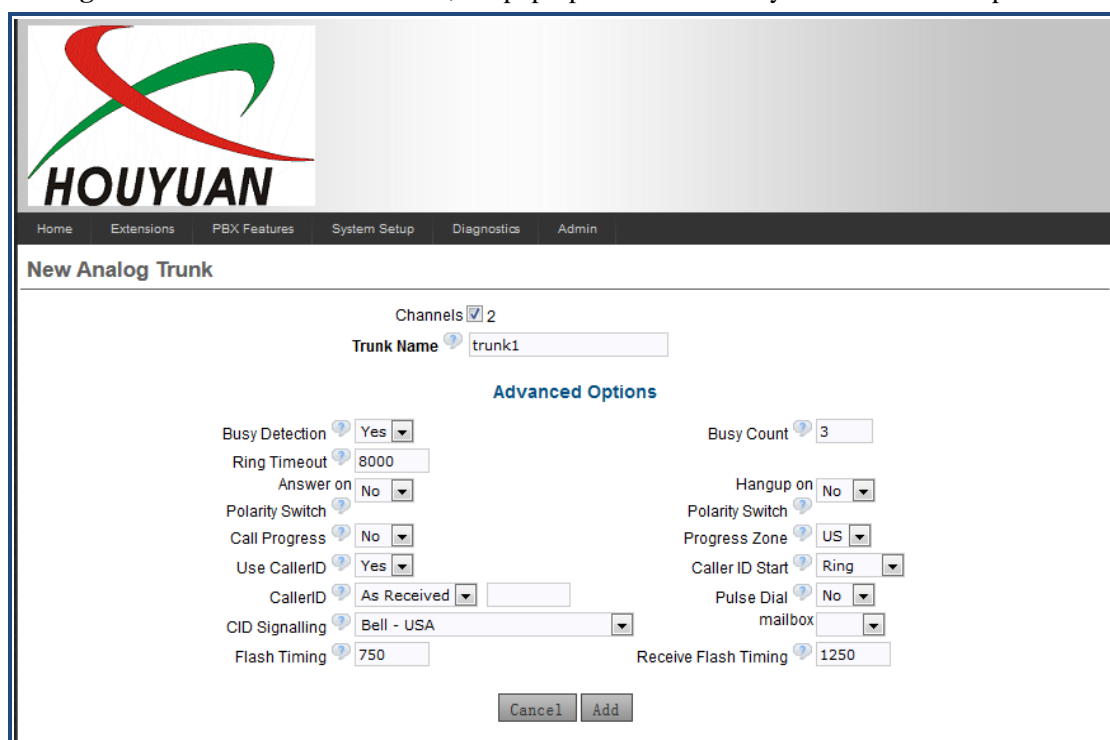
3.3 Trunks

To receive calls from PSTN and make calls to the outside world, you have to use trunk. Please select the **Trunks** option from the vertical menu on the left of the main page, then you can get the following screen:



3.3.1 Create Analog Trunks


Analog trunk is associated with FXO port, and it will call outside by PSTN line. Click on **New Analog Trunk** in the illustration above, the pop-up screen is where you create and set up trunk.



There are many parameters for you to set up, I just set the following two parameters:

Channels: select the FXO port you want to use. Here I use the port 2.

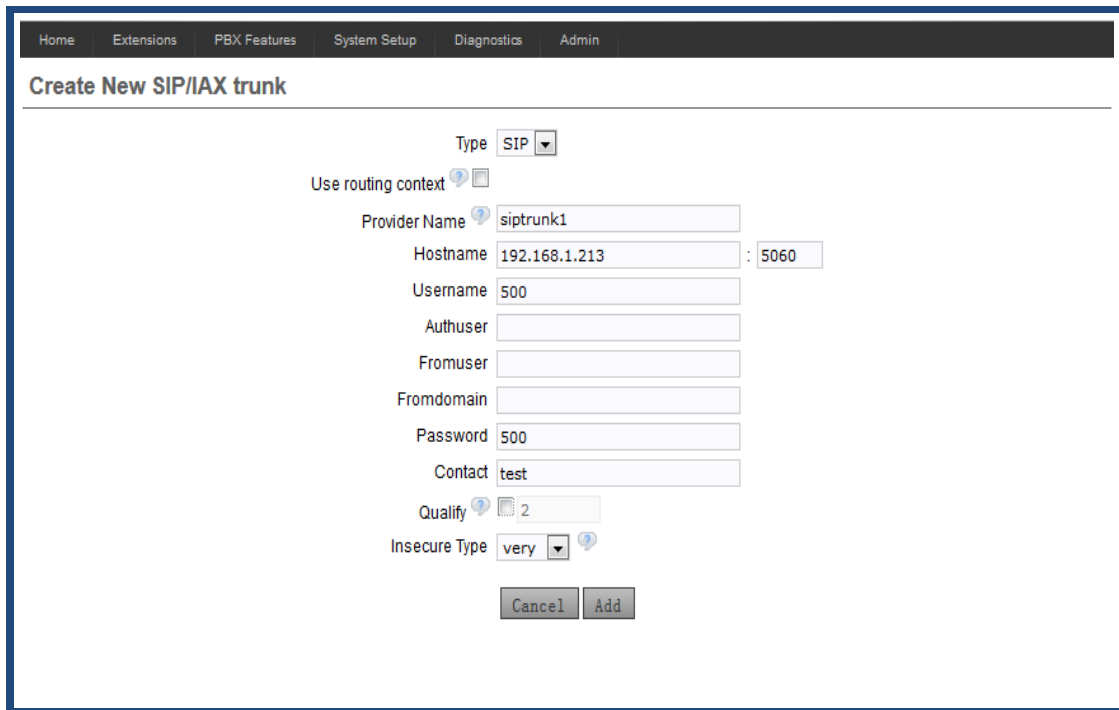
Trunk Name: a unique label to help you identify the trunk when listed in outgoing calling rules and incoming calling rules. Here I use the trunk1 as my trunk name.

For the advanced options, you can put your cursor on the  label, you can get the information of the parameter, customers have to set these parameters according to your service provider and your need.

3.3.2 VoIP Trunks

A VoIP service provider (VSP) that you have signed up with is also a trunk. Via the VoIP trunk you can dial via the VoIP service to reduce your cost when making international calls. You can set up

the VoIP trunk to make calls to the PSTN or other VoIP network depends on the service you use. You can also use the VoIP trunk to link headquarter and branch offices for free internal calls. Click on **New SIP/IAX Trunk**, the following screen is where you create and set up VoIP trunk:



The important parameters are:

Type: You can select SIP or IAX type to meet your need.

Provide Name: a unique label to help you identify the trunk when listed in outgoing calling rules and incoming calling rules.

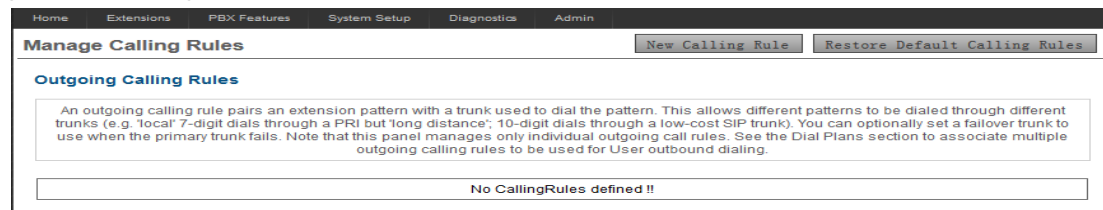
Hostname: the IP address or domain name of your service provider's server.

Username: the username that your service provider configured.

Password: the password that your service provider configured for the user.


3.4 Outgoing Calling Rules


Outgoing calling rules is used to route an outgoing call, when you make an external call, which trunk and what dial-pattern the call used are configured in outgoing calling rules. Please select the Outgoing Calling Rules option from the vertical menu on the left of the main page, then you can get the following screen:





Click on **New Calling Rule** button on the illustration above, the following screen is where you create and set up outgoing calling rule:

New CallingRule



Calling Rule Name  outgoing1


Pattern  _2x.


☒ Send to Local Destination 


Destination 



Send this call through trunk


Use Trunk  trunk1  ☐ Record Calls


Strip  1 digits from front

and Prepend these digits  before dialing

☐ Use FailOver Trunk 

fail over Trunk  

Strip  digits from front

and Prepend these digits  before dialing


The important parameters I configured are below:

Calling Rule Name: a unique label to help you identify the outgoing calling rule when listed in dial plans, I use outgoing1 as the calling rule name here.

Pattern: it acts like a filter for marching numbers you dialed, here I set up _2X., it means any number you dial out with prefix 2 will use this outgoing call rule.

Use Trunk: select the trunk for outgoing calling rule, here I select the trunk1 I set up before.

Strip: I press 1 here, it will strip the first digit of the number string you dialed.

You can get the detail information about every single parameter by putting your cursor on the  label.

At last, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

The way of outgoing calling rules works:

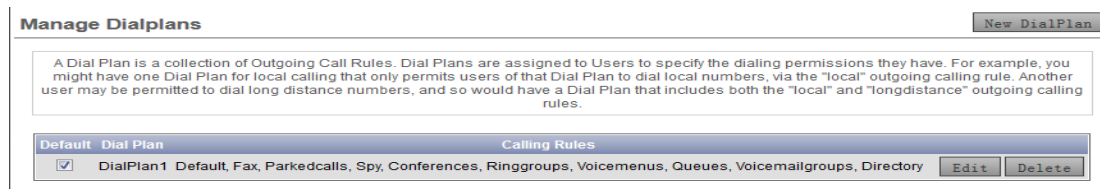
Every time you dial a number, asterisk will do the following in strict order:

- Examine the number you dialed.
- Compare the number with the pattern that you have defined in your first outgoing rule and if matches, it will initiate the call using that trunk. If it does not match, it will compare the number with the pattern that you have defined in the second outgoing rule and so on.
- Pass the number to the appropriate trunk to make the call.

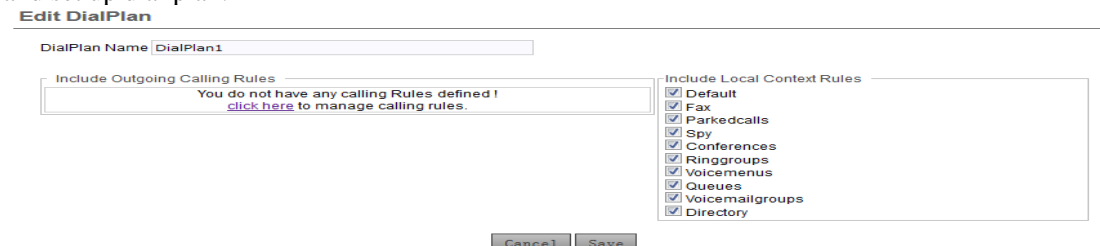
3.5 Dial Plans

A DialPlan is a set of Calling Rules that can be assigned to one or more users. Please select the

Dial Plans option from the vertical menu on the left of the main page, then you can get the following screen:



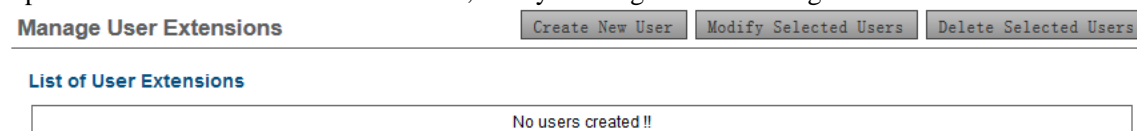
Click on **New DialPlan** button on the illustration above, the following screen is where you create and set up dial plan:



DialPlan Name: a unique label to help you identify the dial plan when listed in user component, you have to set up a dial plan name and select outgoing call rule and local context that you want to use.

3.6 Users

Users component is used to add or remove Analog, SIP, IAX extension. Please select the **Users** option from the vertical menu on the left, then you can get the following screen:



3.6.1 Create SIP/IAX User

Click on **Create New User** button on the illustration above, the following screen is where you create and set up user:

Create New User

General			
Extension ?	6001	Name ?	6001
DialPlan ?	DialPlan1		
Internal CallerID ?	6001	External CallerID ?	6001
<input checked="" type="checkbox"/> Enable Voicemail for this User ?			
Access PIN code ?		Mailbox ?	6001
Email Address ?			
Technology			
SIP <input checked="" type="checkbox"/> IAX <input checked="" type="checkbox"/>	Call Token Required <input type="checkbox"/>	Analog Station ?	None
		flash ?	750
		rxflash ?	1250
Codecs			
First	a-law	Second	u-law
Third	GSM	Fourth	None
Fifth	None		
VoIP Settings			
Qualify <input checked="" type="checkbox"/> NAT <input checked="" type="checkbox"/>	Can Reinvite <input type="checkbox"/>	DTMF Mode ?	RFC2833
		insecure ?	no
		SIP/IAX Password ?	
Other Options			
<input type="checkbox"/> 3-Way Calling ?	<input type="checkbox"/> In Directory ?	<input type="checkbox"/> Call Waiting ?	<input type="checkbox"/> CTI ?
<input type="checkbox"/> Is Agent ?	Pickup Group	1	Call Group
		1	1
<div> <div>Cancel</div> <div>Update</div> </div>			

In General component, you have to set up Extension, CallerID, Name, OutBound CallerID parameters, and choose a DialPlan for the extensions. Here I set up user 6001, and select DialPlan1 for the user.

I select **Enable Voicemail for this User option**, so the user has voicemail function.

In the Technology component, you have to select SIP or IAX. Here I want to configure a SIP user, so I select SIP. For the Codec Preference, only the first two types of code you set are available.

In the **Other Options** component, I select **Is Agent** which will be listed in Call Queues as a selectable member for call queue.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3.6.2 Create Analog User

Click on **Create New User** button, the following screen is where you create and set up user:

Create New User

General			
Extension	6002	Name	6002
Internal CallerID	6002	External CallerID	6002
		DialPlan	DialPlan1
<input checked="" type="checkbox"/> Enable Voicemail for this User			
Access PIN code		Mailbox	6002
		Email Address	
Technology			
SIP	<input checked="" type="checkbox"/>	IAX	<input checked="" type="checkbox"/>
Call Token Required	<input type="checkbox"/>		
Analog Station	Port 1	flash	750
		rxflash	1250
Codecs			
First	a-law	Second	u-law
Third	GSM	Fourth	None
Fifth	None		
VoIP Settings			
Qualify	<input checked="" type="checkbox"/>	NAT	<input checked="" type="checkbox"/>
Can Reinvite	<input type="checkbox"/>		
DTMF Mode	RFC2833	insecure	no
		SIP/IAX Password	
Other Options			
3-Way Calling	<input type="checkbox"/>	In Directory	<input type="checkbox"/>
Call Waiting	<input type="checkbox"/>	CTI	<input type="checkbox"/>
Is Agent	<input type="checkbox"/>	Pickup Group	1
		Call Group	1
Cancel		Update	

In the General component, you have to setup Extension, CallerID, Name, OutBound CallerID parameters, and choose a dialplan for the phone. Here I set up user 6002, and select DialPlan1 for the user.

I select **Enable Voicemail for this User option**, so the user has voicemail function.

In the Technology componet, you have to select the port in which the analog phone will be plugged from the drop-down list of **Analog Station**. I select **Enable Voicemail for this User option**, so the user have voicemail function.

In the **Other Options** component, I select **Is Agent** which will be listed in Call Queues as a selectable member for call queue.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

Attension: in the textbox of Extension, the value you set is limited to a range, you can adjust the range in the following screen to meet your requirement. Please select the **Options** option from the vertical menu on the left, then you can get the following screen:

General Preferences

General Preferences Language Settings Change Password Reset Configuration Reboot Recording Settings

DHCP Server

Global OutBound CID

Operator Extension

Internal Ring Timeout

Outbound Ring Timeout

Extension preferences

User Extensions to

Conference Extensions to

VoiceMenu Extensions to

RingGroup Extensions to

Queue Extensions to

VoiceMail Group Extensions to

Fax2email Extensions to

Reset to defaults

Cancel Save

3.7 Ring Groups

Define Ring groups to dial more than one extension simultaneously, or to ring more than one phone sequentially. This feature may also be called Hunt groups.

Please select the **Ring Groups** option from the vertical menu on the left of the main page, then you can get the following screen:

Manage RingGroups New RingGroup

No RingGroups defined !!

Click on **New RingGroup** button on the illustration above, the following screen is where you create and set up ring group:

New RingGroup

RingGroup Name

Extension for this ring group

Ring Group Members

6003(SIP) 6003
6002(SIP) 6002

Available Users

6001(SIP) 6001
6001(IAX2) 6001

<<<
←
→
>>>

Ring Group Options

Strategy Ring in Order

Seconds to ring each member 20

If not answered Goto Hangup

Cancel
Save

Set the ring group name and extension for the ring group, select ring group members from available users.

Select strategy for ring group:

Ring in Order: when someone calls the ring group, the ring group member will ring in order.

Ring all simultaneously: when someone calls the ring group, all of the ring group member will ring at the same time.

If not answered Goto: choose a destination from the drop-down list, when no one in the ring group answers the call.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3.8 Call Queues

Please select the **Call Queues** option from the vertical menu on the left of the main page, then you can get the following screen:

Queues
Create New Queue

Queues
Agent Login Settings

No Call Queues defined !!

Click on **Create New Queue** button on the illustration above, the following screen is where you

create and set up call queue:

Edit Queue 6500

Extension

Name

Strategy

Music On Hold

LeaveWhenEmpty

JoinEmpty

Hold TimeOut

Queue Options

TimeOut Wrapup Time Max Len

Auto Fill ☐ Auto Pause ☐ Report Hold Time ☐

KeyPress Events

☐ Enable initial Anouncement

Wait Before Wait After

☐ Periodic Announcement
 Frequency (Sec)

☐ Enable Exit to

Agents

☒ 6002 (6002) ☒ 6003 (6003)

Members

☐ SIP/6001
☐ IAX2/6001
☒ SIP/6002
☒ SIP/6003
☐ DAHDI/1

Cancel

Update

Extension: a unique label to help you identify the call queue when listed in **outgoing calling rules** component.

Agents: select the users which you want them to be queue member.

You can get information of other parameters by putting your mouse on the  label.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3.9 Voice Menus

Like most organization, we would like to redirect all of the incoming calls automatically. The voice menu is very handy for these sorts of things. The system should allow callers to make the selection according to the voice menu.

Please select the **Voice Menus** option from the vertical menu on the left, then you can get the following screen:

Manage Voice Menus

Create New VoiceMenu

No VoiceMenus defined !!

Click on **Create New VoiceMenu** button on the illustration above, the following screen is where you create and set up voice menu:

Edit VoiceMenu voicemail-custom-1 Advanced Edit

General Key Press Events

Name:

Extension:

☒ Allow Dialing Other Extensions

Actions

Answer the call	↓ ↑ ×
Play 1-for-am-2-for-pm & Donot Listen for KeyPress events	↓ ↑ ×
Goto User 6001	↓ ↑ ×

Add New Action

Cancel Save

Key Press Events

General Key Press Events

Key	Action	Key	Action
0	Goto Operator	8	--
1	Goto RingGroup ringgroup1	9	--
2	Goto User 6001	#	--
3	--	*	--
4	--	t	--
5	--	i	--
6	--	fax	--
7	--		

Name: a unique label to help you identify the voice menu when listed in incoming calling rules.

Add new Step: select an action from the drop-down list. I add three steps above, so it will answer the call, and play a sound file, at last go to user 6001.

Click on **Allow KeyPress Events**: when the caller is in voice menu, they can press some specific numbers which are defined here to enter other destination. Here I define three numbers for going to operator, ringgroup, and user respectively.

At last, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

3.10 Time Intervals

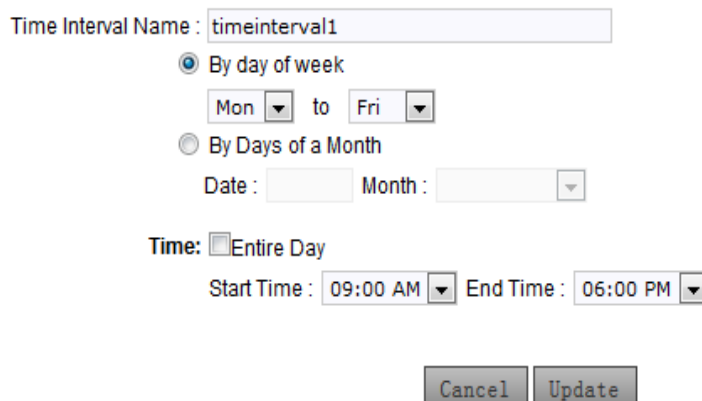
Time Intervals defines ranges of working time that will be used by call routing features. Please select the **Time Intervals** option from the vertical menu on the left of the main page, then you can get the following screen:

Time Intervals New Time Interval

No Time Intervals defined !!

Click on **New Time Interval** button on the illustration above, the following screen is where you create and set up time interval:

New Time Interval



The form for creating a new time interval. It includes a text field for 'Time Interval Name' with the value 'timeinterval1'. There are two radio buttons: 'By day of week' (selected) and 'By Days of a Month'. The 'By day of week' option has dropdowns for 'Mon' and 'Fri' with 'to' in between. The 'By Days of a Month' option has 'Date' and 'Month' dropdowns. There is a checkbox for 'Time' labeled 'Entire Day' which is unchecked. Below that are 'Start Time' and 'End Time' dropdowns showing '09:00 AM' and '06:00 PM' respectively. At the bottom are 'Cancel' and 'Update' buttons.

Time Interval Name: a unique label to help you identify the time interval when listed in incoming calling rules. I set up timeinterval1 as time interval name.

By day of week: I select it from Monday to Friday, the incoming call rule only works from Monday to Friday.

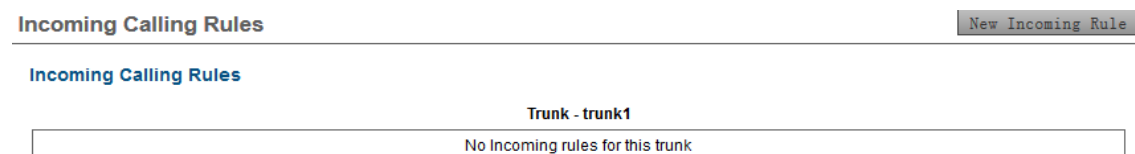
Time: I set up it from 09:00 AM to 06:30 PM, the incoming call rule only works from 09:00 AM to 06:30 PM.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3.11 Incoming Calling Rules

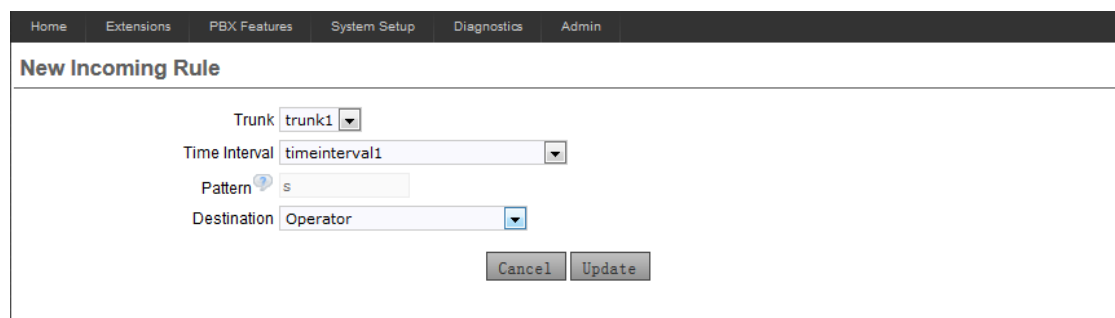
This is where the behavior of incoming calls from all trunks is being handled. When an incoming call from PSTN or VoIP trunk is received, asterisk needs to know where to direct it. It can be directed to a ring group, an extension, digital receptionist, voice menu or queue. For this purpose, Incoming Calling Rules need to be set up.

Please select the **Incoming Calling Rules** option from the vertical menu on the left of the main page, then you can get the following screen:



The screen for Incoming Calling Rules. It has a header with 'Incoming Calling Rules' on the left and 'New Incoming Rule' on the right. Below the header is a sub-header 'Incoming Calling Rules'. In the center, there is a box labeled 'Trunk - trunk1' containing the text 'No Incoming rules for this trunk'.

Click on **New Incoming Rule** button on the illustration above, the following screen is where you create and set up time interval:



Trunk: select trunk for incoming call to use. I select trunk1 I set up before.

Time Interval: determine the time when the incoming call rule works, I select timeinterval1 I set up before.

Pattern: match the destination number, I use S which will match any destination number.

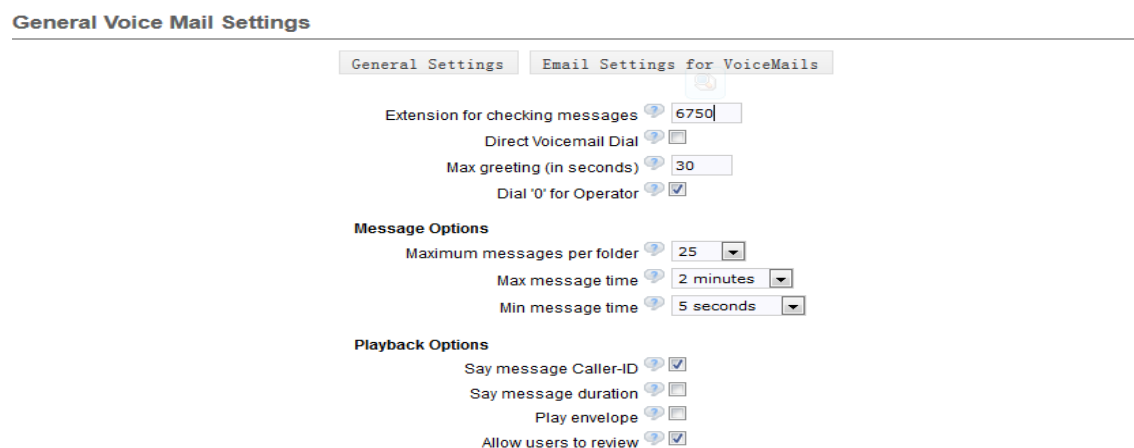
Destination: I select Operator, so the call will be ruled to Operator.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3.12 Voicemail

When you call someone who does not answer the call, you can leave a voice message for the called party if the called party supports voice mail.

Please select the **Voicemail** option from the vertical menu on the left of the main page, then you can get the following screen:





Click on **General Settings** button on the illustration above. You can see the following screen:


General Voice Mail Settings


General Settings


Email Settings for VoiceMails





Extension for checking messages  6750



Direct Voicemail Dial  ☐



Max greeting (in seconds)  30

Dial '0' for Operator  ☒


Message Options


Maximum messages per folder  25 


Max message time  2 minutes 


Min message time  5 seconds 

Playback Options


Say message Caller-ID  ☒

Say message duration  ☐

Play envelope  ☐

Allow users to review  ☒

Extension for checking messages: when you dial 6750, you will hear the voice message other people left for you.

You can get information of parameters by putting your cursor on the  label. If you want to set voicemail function for the user, you have to enable voicemail component when you set up a user. Please refer to the following illustration:

Edit User Extension ! - 6001

General		
Extension ? 6001	Name ? 6001	DialPlan ? DialPlan1
Internal CallerID ? 6001	External CallerID ? 6001	
<input checked="" type="checkbox"/> Enable Voicemail for this User ?		
Access PIN code ?	Mailbox ? 6001	Email Address ?
Technology		
SIP <input checked="" type="checkbox"/> ? IAX <input type="checkbox"/> ? Call Token Required <input type="checkbox"/>	Analog Station ? None <input type="checkbox"/> flash ? <input type="checkbox"/> rxfash ? <input type="checkbox"/>	
Codecs		
First a-law <input type="checkbox"/> ? Second u-law <input type="checkbox"/> ? Third GSM <input type="checkbox"/> ? Fourth None <input type="checkbox"/> ? Fifth None <input type="checkbox"/> ?		
VoIP Settings		
Quality <input checked="" type="checkbox"/> ? NAT <input checked="" type="checkbox"/> ? Can Reinvite <input type="checkbox"/> ? DTMF Mode ? RFC2833 <input type="checkbox"/> insecure ? very <input type="checkbox"/> ? SIP/IAX Password ?		
Other Options		
<input type="checkbox"/> 3-Way Calling ? <input type="checkbox"/> In Directory ? <input type="checkbox"/> Call Waiting ? <input type="checkbox"/> CTI ? <input checked="" type="checkbox"/> Is Agent ? Pickup Group 1 <input type="checkbox"/> Call Group 1 <input type="checkbox"/>		
<div>Cancel Update</div>		

3.13 Conferencing

The conferencing function of Asterisk is similar to a Tele-conference call where multiple callers can call in and participate in a two-way conference like in a party room where everyone can talk and listen to one another or just to listen to a Tele-presentation.

Please select the **Conferencing** option from the vertical menu on the left of the main page, then you can get the following screen:

Manage Conference Rooms

[New Conference Bridge](#)

No Conference rooms defined !!

Click on **New Conference Bridge** button on the illustration above. Below is what my conference configuration page looks like:

New Conference Bridge

Extension
Marked/Admin user Extension

Password Options

Pin Code
Admin PinCode


Conference Room Options

☒ Play hold music for first caller
☐ Close conference when last marked user exits

☐ Enable caller menu
☒ Announce callers

☐ Quiet Mode
☐ Wait for marked user

Cancel Update

Naturally there are some options that you may wish to have for the conference room. They are entirely up to you. The main important things are for you to create the conference room number and the conference pin code for you to know how to enter into the conference. The rest of the fields are optional. You can get information of other parameters by putting your mouse on the  label.

This conference number is 6300, the Pin Code is 123 for common member, the Pin Code is 456 for Admin. So you have to dial 6300 then, press the Pin Code, if you want to enter the conference. I enable the play hold music for option and announce callers option, so the first member who enter the conference will listen to a music and the online members will be informed when someone enter the conference.

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3.14 Follow Me

If A calls B, B does not answer, the call will be transferred to C who is set up in follow me.

Please select the **Follow Me** option from the vertical menu on the left, then you can get the following screen:

Follow Me Preferences for Users

Follow Me Preferences for Users		Follow Me Options
Extension	Follow Me	Follow Order
6001	Disabled	Not Configured
6002	Disabled	Not Configured
6003	Disabled	Not Configured

You can choose user for which you want to setup follow me function, Here taking the user 6006

for an example, click on the **edit** button at the same line as 6006, you can get the following screen:

Edit User 6006

Status ☒ Enable ☐ Disable

'Music On Hold' Class

DialPlan

Destinations

Add Follow Me Number

Cancel Save

Select the **enable** status, and click on **Add FollowMe Number** button to add a destination phone.

Edit User 6006

Status ☒ Enable ☐ Disable

'Music On Hold' Class

DialPlan

Destinations

New FollowMe Number ☒ Dial Local Extension ☐ Dial Outside Number

Dial Order for 30 Seconds

6002 6002 g previous extension/number

6003 6003 n previous extension/number

6006 6006

Cancel Add

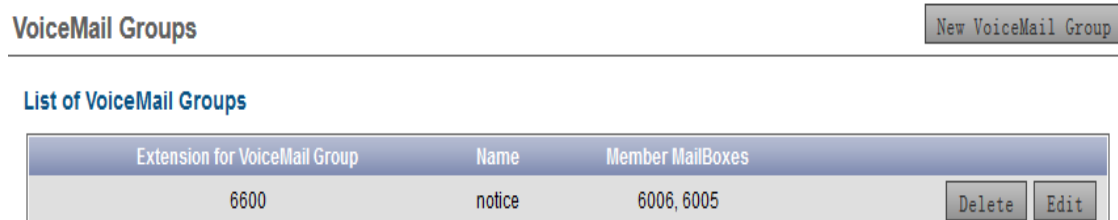
Click on **Dial Local Extension** and select 6001. Click on **Add** button and click on **Apply Changes** button in up right corner of the main page.

Through the above settings, someone calls 6006, but 6006 does not answer, the call will be transferred to 6001 automatically.

3.15 VoiceMail Groups

Define VoiceMail Groups to leave a voicemail message for a group of users by dialing an extension.

Please select the **VoiceMail Groups** option from the vertical menu on the left of the main page, then you can get the following screen:



Extension for VoiceMail Group	Name	Member MailBoxes	
6600	notice	6006, 6005	Delete Edit

Click on **New VoiceMail Group** button on the illustration above. Below is what my VoiceMail Group configuration page looks like:



Edit Voice Mail Group - 6600

VoiceMail Group's Extension: 6600

Name: notice

User MailBoxes: ☐ 6001 ☐ 6002 ☒ 6006 ☒ 6005

Cancel Save

From the above settings, I can dial 6600 to leave message for user 6005 and 6006.

3.16 Voice Menu Prompts

This component is used for recording custom voice menu.

Please select the **Voice Menu Prompts** option from the vertical menu on the left of the main page, then you can get the following screen:

Custom Voice Menu Prompts

[Delete Selected](#)[Record a new Voice Menu prompt](#)[Upload a Voice Menu prompt](#)

List of Custom Voice Menu Prompts

No custom Voice Menu prompts found !!

You can record a new VoiceMenu Prompt by clicking on the 'Record a new Voice Menu prompt' or click on the 'Upload a Voice Menu prompt' button to upload a custom voice menu.

Click on **Record a new Voice Menu prompt** button on the illustration above. Below is what my Record a new Voice Menu prompt configuration page looks like:

Record a new Voice Menu prompt

File Name

Welcome

Dial this User Extension to record a new voice prompt

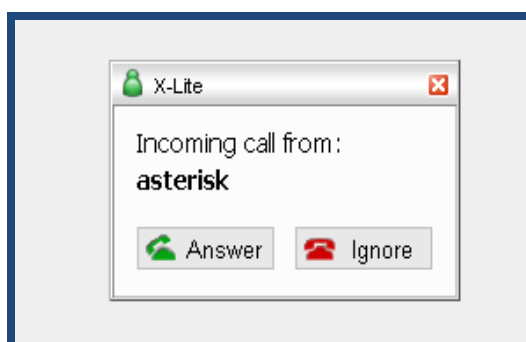
GSM
GSM
a-Law
u-Law
G.729

[Cancel](#)[Record](#)

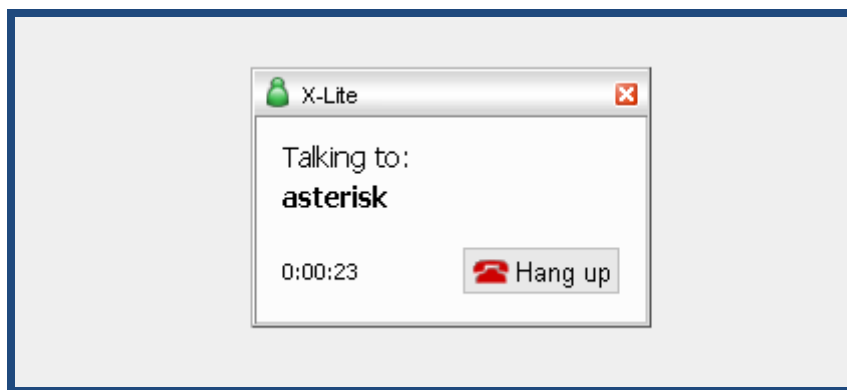
File Name: give a filename for the record sound file, here I give a name: Welcome

Dial this User Extension to record a new voice: dial to a user, then the user pick up the phone and speak the voice menu which will be recorded. Here I select 6001 I set up before.

Click on **Record** button, the asterisk will call to 6001, 6001 will show like the following:



Click on **Answer** button, then you call speak and start to record what you say. The following illustration will be presented after you click on the **Answer** button.



When you want to finish the record, please click on **Hang up** button.

[Record a new Voice Menu prompt](#) [Upload a Voice Menu prompt](#)

List of Custom Voice Menu Prompts

<input type="checkbox"/>	Name	<input type="checkbox"/>
<input type="checkbox"/>	Welcome.gsm	<input type="checkbox"/>

After you finish the recording, please refresh you webpage, and enter into **voice menu prompts** component again, you can see you have had a sound file like the above.

Network Information:

Networking setting

WAN Interface DHCP <input type="text" value="no"/> Hostname <input type="text" value="ip04"/> Domain <input type="text" value="switchfin.org"/> MAC <input type="text" value="00:16:D3:2A:C5:78"/> IP address <input type="text" value="192.168.1.100"/> Subnet mask <input type="text" value="255.255.255.0"/> Gateway <input type="text" value="192.168.1.1"/> DNS <input type="text" value="192.168.1.1"/> NTP <input type="text" value="pool.ntp.org"/>	VLAN Interface for WAN VLAN <input type="checkbox"/> Vlan number <input type="text" value="100"/> Vlan IP address <input type="text" value="192.168.100.100"/> Vlan Subnet mask <input type="text" value="255.255.255.0"/> Vlan Gateway <input type="text" value="192.168.100.1"/>
LAN Interface Show LAN Settings <input type="checkbox"/> IP address <input type="text"/> Subnet mask <input type="text"/> MAC <input type="text"/>	DynDNS Enable DynDNS <input type="checkbox"/> Username <input type="text"/> Password <input type="text"/> Domain <input type="text"/>
System TimeZone TimeZone <input type="text" value="America/New_York"/>	

Disk Usage Information:

Uptime: 6 min

CPU Usage:

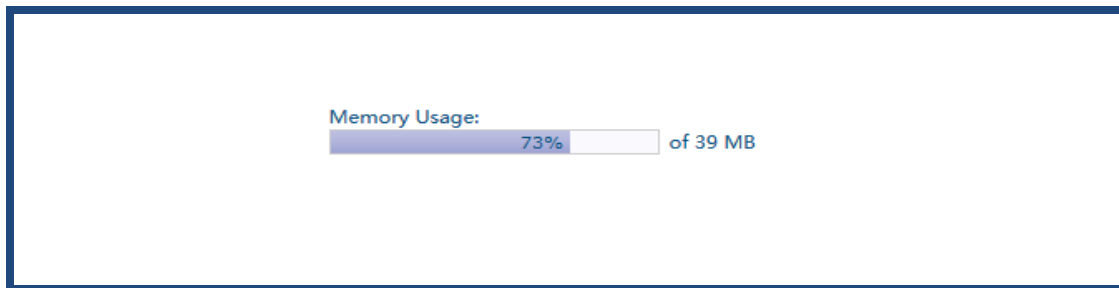
Memory Usage:
 of 39 MB

Root Filesystem Usage:
 of 17 MB

Persistent Filesystem Usage:
 of 213 MB



Memory Usage Information:



3.18 Backup

Backup and Restore are two of the mandatory functions of any application. IPOx is no exception. Customers can backup all the files under the /etc/asterisk/ directory and restore them.

Please select the **Backup** option from the vertical menu on the left of the main page, then you can get the following screen:

Backup / Restore Configurations

Create New Backup

Upload Backup File

List of Previous Configuration Backups

No Previous Backup configurations found !!

Please click on the 'Create New Backup' button to take a backup of the current system configuration

Click on **Create New Backup** button on the illustration above, you can get the following illustration:

Backup / Restore Configurations

Create New Backup

Upload Backup File

List of Previous Configuration Backups

#	Name	Date	
1	backup_2013aug28_083852	Aug 28, 2013	Download from Unit Restore Delete

File Name: give a file name for the backed up file.

Click on **Backup** button, once the backup process is completed, you will see a screen with the backup filename displayed in illustration below.

Backup / Restore Configurations[Create New Backup](#)[Upload Backup File](#)**List of Previous Configuration Backups**

#	Name	Date	
1	backup_2013aug28_083852	Aug 28, 2013	Download from Unit Restore Delete


Backup itself is not useful if it cannot be restored, IP0x also has this function. This is a very simple procedure. All you need to do is to click on the **Restore Previous Config** option.

3.19 Active Channels

The channels which are in communication status will be displayed in this component.

Please select the **Active Channels** option from the vertical menu on the left, then you can get the following screen:

Operator Panel | Admin | [Logout](#)



Home | Extensions | PBX Features | System Setup | Diagnostics | Admin

Channel Management [Refresh Now](#)

Refreshing Active Channels in 1 Seconds

No Channels Open !!

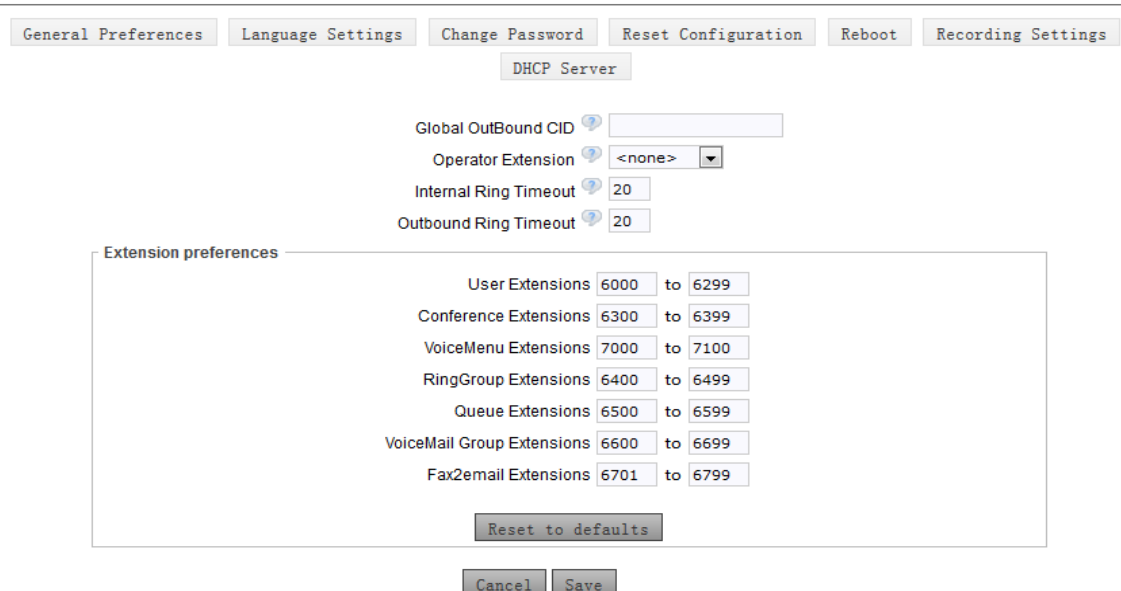
3.20 Options

This component is used for administrator to manage the system, it includes the following modules:

- General Preferences
- Language
- Change Password
- Factory Reset
- Reboot

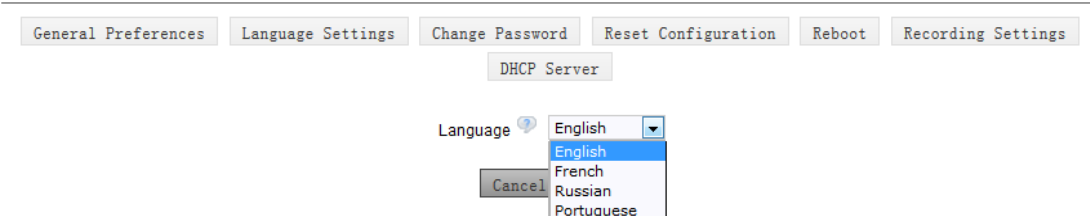
General Preferences: you can set up a user to be the operator and the range of extension number for different types' extensions like the following screen:

General Preferences



Language: change the sound file language in which they play.

Language preferences



Change Password: it is used for customers to change the admin password, click on the **Change Password** button, the following illustration will be presented below:

Change Password

General Preferences Language Settings **Change Password** Reset Configuration Reboot Recording Settings

DHCP Server

Enter New Password

Retype New Password

Update

After inputting your new password, please click on **Update** button, then click on **Apply Changes** button on the up right corner of the main page.

Factory Reset: it will help you to recover to the default factory settings. Click on **Factory Reset** button, the following illustration will be presented below:

Reset to Factory Defaults

General Preferences Language Settings Change Password **Reset Configuration** Reboot Recording Settings

DHCP Server

Warning: By resetting your PBX Appliance/System to factory defaults, you will lose all your configuration !
You can take a backup of your current configuration from the [Backup page](#).

☒ Keep Network Settings

Reset to Defaults

Please click on **Reset to Defaults** button to recover to default factory setting, then click on **Apply Changes** button on the up right corner of the main page.

Reboot: you can click on **Reboot** button→**Reboot Now** button to reboot your system.

Recording Setting:

General Preferences

General Preferences Language Settings Change Password Reset Configuration **Reboot** Recording Settings

DHCP Server

SPY (*779) Password

WHISPER (*9447737) Password

Recording Format *.wav

Cancel Save

DHCP Server Options

General Preferences
Language Settings
Change Password
Reset Configuration
Reboot
Recording Settings

DHCP Server

Standard Settings

Enable DHCP Server ☐

Interface eth0

Start IP 192.168.1.101

End IP 192.168.1.199

Max Leases 20

Options

DNS IP 192.168.1.1

Subnet Mask 255.255.255.0

Router IP 192.168.1.1

Domain Name switchfin.org

Lease Time 864000

Cancel Save

3.21 Asterisk Logs

After click on **Diagnostics>PBX Log messages**, please select the **PBX Logs** option from the vertical menu on the left of the main page, then you can get the following screen:

HouYuan PBX Log messages
Select Date: Go

Disable Logger
☐ notice
☐ warning
☐ error
☐ debug
☐ verbose

Click on the textbox, you can get the following screen:

PBX Log messages
Select Date: 9 Aug 2013 Go

Logger
☐ notice
☐ warning
☐ error
☐ debug
☐ verbose

August 2013

Mon	Tue	Wed	Thu	Fri	Sat	Sun
29	30	31	1	2	3	4
5	6	7	8	9	10	11
12	13	14	15	16	17	18
19	20	21	22	23	24	25
26	27	28	29	30	31	1

You can see a date table, and you can select the log to watch by clicking on the date. After choosing the date, please click on **Go** button, you can see the asterisk log of the day you choosed. Here I need to see the asterisk log of August 9st, 2013, I click on 9 in the date table, I get the following screen:

Select Date: 9 Aug 2013
Go

I click on **Go** button, then I get the log in the following screen:

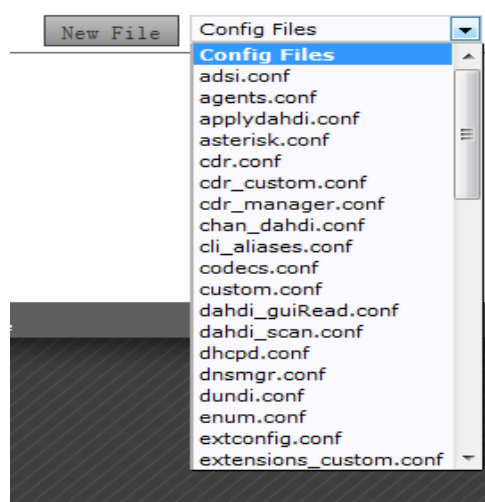

```
[Apr 21 03:44:29] WARNING[19672] chan_sip.c: Ignoring insecure
[Apr 21 03:44:29] WARNING[19672] chan_sip.c: Ignoring signalling
[Apr 21 03:44:29] WARNING[19672] chan_sip.c: Ignoring macaddress
[Apr 21 03:44:29] WARNING[19672] chan_sip.c: Ignoring outprov
[Apr 21 03:44:29] WARNING[19672] chan_sip.c: Ignoring label
[Apr 21 03:44:29] WARNING[19672] chan_sip.c: Ignoring linenumber
[Apr 21 03:44:29] WARNING[19672] chan_sip.c: Ignoring flash
[Apr 21 03:44:29] WARNING[19672] chan_sip.c: Ignoring disallow
[Apr 21 03:44:29] WARNING[19672] chan_sip.c: Ignoring allow
[Apr 21 03:45:16] WARNING[19890] app_dial.c: Unable to create channel of type 'IAX2' (cause 3 - No route to destination)
[Apr 21 03:45:36] NOTICE[211] chan_sip.c: -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #1)
[Apr 21 03:45:40] WARNING[19891] ast_expr2.fl: ast_yyerror(): syntax error: syntax error, unexpected 'u', expecting $end: Input:
[Apr 21 03:45:40] WARNING[19891] ast_expr2.fl: If you have questions, please refer to doc/channelvariables.txt in the asterisk source.
[Apr 21 03:46:06] WARNING[19891] app_dial.c: Unable to create channel of type 'IAX2' (cause 3 - No route to destination)
[Apr 21 03:46:26] NOTICE[211] chan_sip.c: -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #2)
[Apr 21 03:47:16] NOTICE[211] chan_sip.c: -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #3)
[Apr 21 03:47:46] WARNING[211] chan_sip.c: Maximum retries exceeded on transmission 24806208277904-200421191943610192.168.1.3 for seqno 1 (Critical Response)
[Apr 21 03:47:46] WARNING[211] chan_sip.c: Hanging up call 24806208277904-200421191943610192.168.1.3 - no reply to our critical packet.
[Apr 21 03:48:06] NOTICE[211] chan_sip.c: -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #4)
[Apr 21 03:48:56] NOTICE[211] chan_sip.c: -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #5)
[Apr 21 03:49:46] NOTICE[211] chan_sip.c: -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #6)
[Apr 21 03:50:36] NOTICE[211] chan_sip.c: -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #7)
[Apr 21 03:51:26] NOTICE[211] chan_sip.c: -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #8)
[Apr 21 03:52:16] NOTICE[211] chan_sip.c: -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #9)
[Apr 21 03:53:06] NOTICE[211] chan_sip.c: -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #10)
[Apr 21 03:53:56] NOTICE[211] chan_sip.c: -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #11)
[Apr 21 03:54:46] NOTICE[211] chan_sip.c: -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #12)
[Apr 21 03:55:36] NOTICE[211] chan_sip.c: -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #13)
[Apr 21 03:56:26] NOTICE[211] chan_sip.c: -- Registration for '5000192.168.1.213' timed out, trying again (Attempt #14)
```

3.23 File Editor

please select the **File Editor** option from the vertical menu on the left, then you can get the following screen:



From the drop-down list of config files, you can select the file you want to edit or read.



Here I select users.conf file, so I can see the file and edit to meet my requirement.

File Editor - users.conf

New File users.conf

Add Context

+ [general]

+ [trunk_1]

+ [6001]

+ [6002]

+ [6003]

+ [6006]

+ [6005]

+ [6000]

3.24 Asterisk CLI

These are some of the available CLI commands that can be executed from the console, you can input the asterisk CLI commands from the web page directly.

please select the

Asterisk CLI option from the vertical menu on the left, then you can get the following screen:

HouYuan PBX CLI	CLI Command: help
Command> help	
<pre> ! Execute a shell command abort halt Cancel a running halt agent logoff Sets an agent offline agent show Show status of agents agent show online Show all online agents agi debug Enable AGI debugging agi debug off Disable AGI debugging agi dumphtml Dumps a list of agi commands in html format agi show List AGI commands or specific help cdr status Display the CDR status core set debug channel Enable/disable debugging on a channel core set debug Set level of debug chattiness core set debug off Turns off debug chattiness core set global Set global dialplan variable core set verbose Set level of verbosity core show applications Shows registered dialplan applications core show application Describe a specific dialplan application core show audio codecs Displays a list of audio codecs </pre>	

Here I input help command in the textbox, so I can get all the command which I can use in CLI mode.

3.25 Network Settings

In order to give a static and permanent IP address for IP0x, you have to set it in web GUI. After you enter into the web GUI of IP0x, you can try to configure IP address according to the following

steps:

please select **Network Settings**

option from the vertical menu on the left of main page, the following screen is where you configure the network:

WAN Interface

DHCP	no ▼
Hostname	ip0x
Domain	switchfin.org
MAC	00:16:D3:2A:C5:78
IP address	192.168.1.100
Subnet mask	255.255.255.0
Gateway	192.168.1.1
DNS	192.168.1.1
NTP	pool.ntp.org

In the drop-down list of **DHCP**, you can see the following three options:

1. DHCP: yes: IP0x will obtain the dynamic IP address from your router.
2. DHCP: auto: IP0x will use the static IP specified below and ping the default gateway. When there is no response from the default gateway, the IP0x will switch to dynamically obtain the IP address from your router.
3. DHCP: no: IP0x will use the static IP address set below.

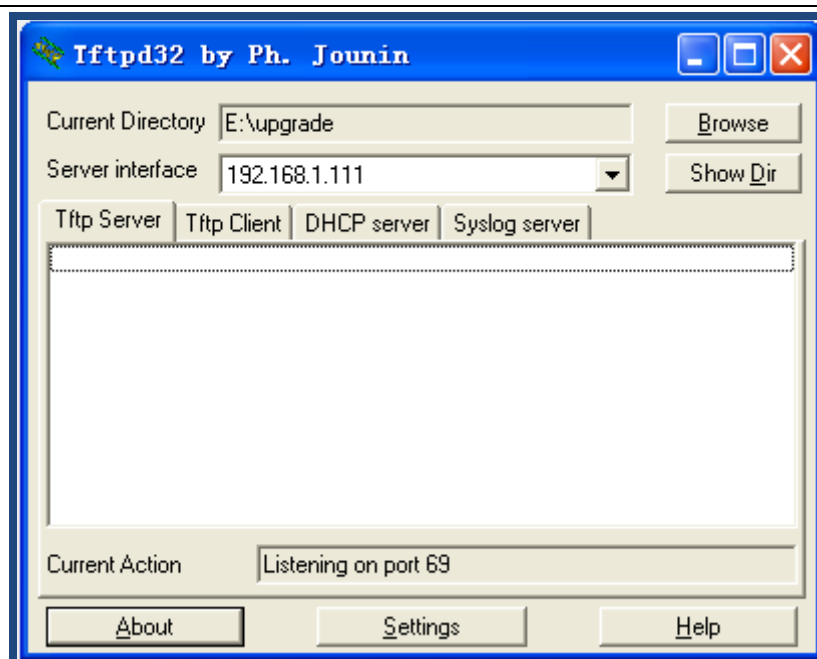
If you want to get static and permanent IP address, please do not select “yes”, after configure other parameters, please click “save” in the bottom of your page to save your setting.

3.26 Firmware Update

You can update to the latest version for IP0x by TFTP.

3.26.1 Download the Latest Firmware File and Set up TFTP Server.

- 1) Download the md5 file from http://www.houyuan.com/downloads/IPPBX/Firmware/IP02_08.md5, then put it in your TFTP server root directory.
- 2) Run your TFTP server, and I set up it like the following:



“E:\upgrade” is the root directory of my TFTP server, “192.168.1.111” is the IP Address of my TFTP server.

3.26.2 Update for IP02 from Web Page

please select **Firmware**

update option from the vertical menu on the left of main page, the following screen is where you update for IP02:

Update Firmware

☐ Web Update
 ☐ HTTP URL
 ☒ TFTP Server

TFTP Server

File Name

☐ Reset Configs
 ☐ Keep Network Settings

TFTP Server: enter the IP Address of your TFTP server in this textbox.

File Name: enter the update file name

Reset Configs: if you choose reset Configs, it will delete all of your configuration you have done before.

After setting up, please click on **Go** button to update for IP02.

Power off and power on the IP02, wait for several minutes. When the TEL port LED light up, it means the update is finished and you have the latest firmware.

3.27 Call Detail Records

This component provides the record of all incoming and outgoing calls including the channels used and duration of calls.

, please select the **Call Detail Records** option from the vertical menu on the left, then you can get the following screen:

CDR Viewer

CDR File Manager
<< prev
next >>
View: 10

☒ Inbound calls
☒ Outbound calls
☒ Internal calls
☒ External calls
☐ Show system calls

	Type	Start time	Duration	Source	Destination	Caller ID	Disposition
1	←	2013-05-09 09:16:12	0:00:08	0000000	s	"unknown" <0000000>	ANSWERED
2	←	2013-05-09 08:38:25	0:00:09	0000000	s	"unknown" <0000000>	ANSWERED
3	←	2013-05-09 08:35:34	0:00:09	0000000	s	"unknown" <0000000>	ANSWERED
4	←	2013-05-09 08:35:06	0:00:09	0000000	s	"unknown" <0000000>	ANSWERED
5	⬆	2013-05-09 08:32:25	0:00:23	0000000	6000	"unknown" <0000000>	NO ANSWER
6	⬆	2013-05-09 08:30:24	0:00:20	0000000	6000	"unknown" <0000000>	NO ANSWER
7	↻	2007-01-01 00:11:16	0:00:21	6000	2006	"6000" <6000>	ANSWERED
8	↻	2007-01-01 00:11:05	0:00:01	6000	2006	"6000" <6000>	ANSWERED
9	↻	2007-01-01 00:05:18	0:00:00	6001	6000	"6001" <6001>	FAILED
10	↻	2007-01-01 00:05:01	0:00:00	6001	6000	"6001" <6001>	FAILED

You can click on the **prev** to look up the last page for call record, and click on the **next** to look up the **next** page for call record, you can also set the value from the drop-down list of **view** which means how many calls will be displayed in one page.

Chapter 4 an Application Case of IP PBX02

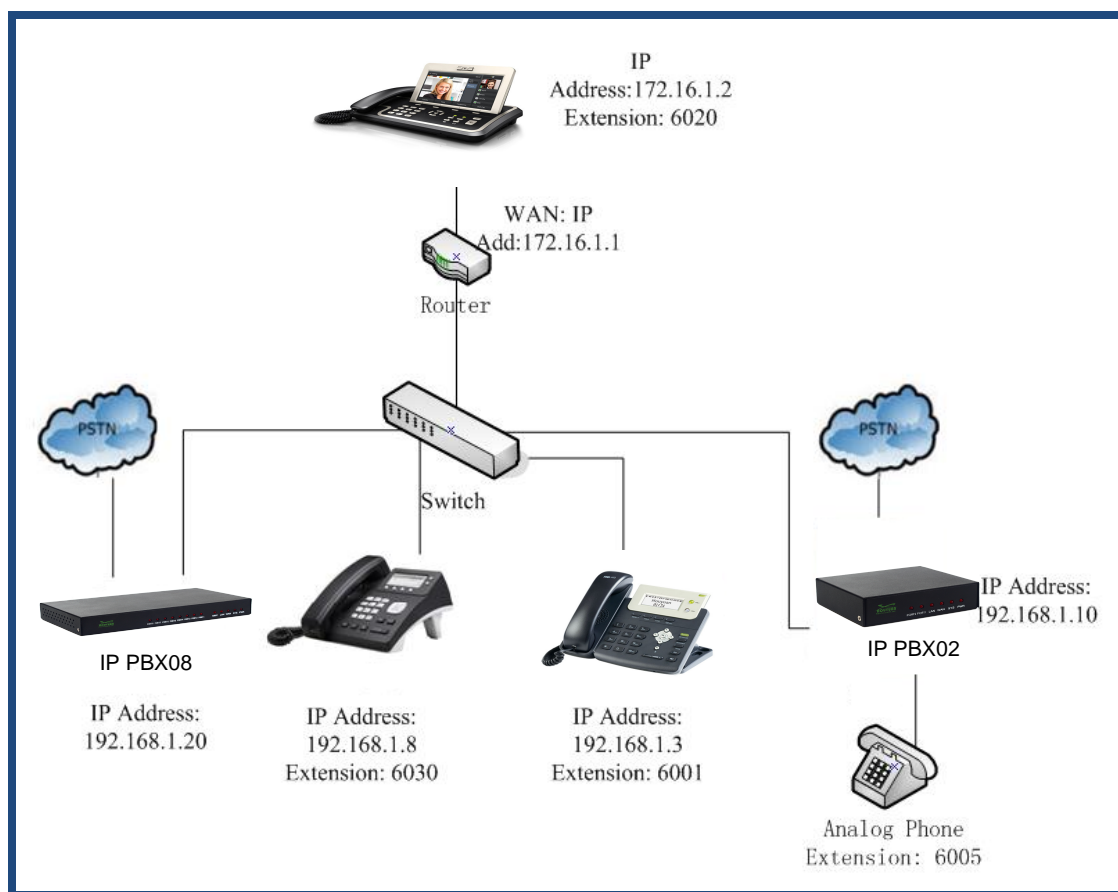


Figure: Network Topology

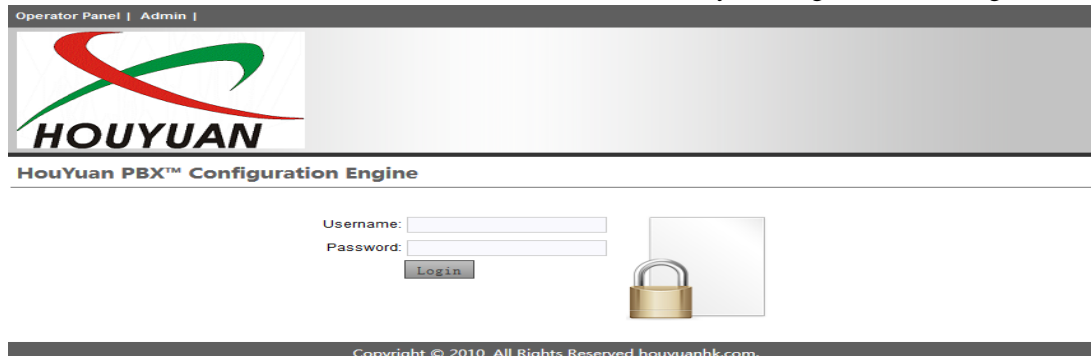
In the network topology above: user 6020 and user 6001 will be registered to IP02, user 6030 will be registered to IP08, analog phone 6005 is connected to FXS port of IP02. After configuration, it will realize the following function:

- 1) The internal user 6005 and user 6001 can call each other directly.
- 2) 6005 and 6001 can dial-out through IP02 to PSTN.
- 3) 6005 and 6001 can get incoming calls from PSTN by IP02.
- 4) 6030 can call-out to PSTN and get incoming call from PSTN through IP08.
- 5) User 6001 and 6030 can call each other through VoIP trunk, although they are registered to different IP PBX.
- 6) User 6020, 6005 and 6001 can call each other directly, although they are not in the same network segment.

4.1 How to Make Internal Calls through IP0x

4.1.1 Access to the Web Page of IP0x by Browser

After connecting IP0x to LAN, please open your browser of PC with windows OS and input the IP Address of IP0x (the default IP address is 192.168.1.100), then you can get the following screen:



Operator Panel | Admin |

HOUYUAN

HouYuan PBX™ Configuration Engine

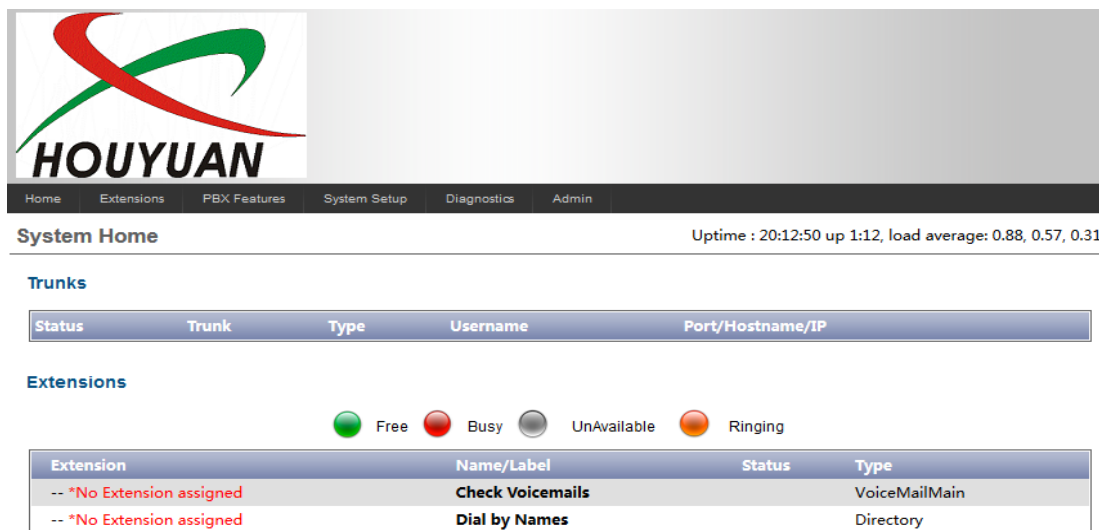
Username:

Password:

Login

Copyright © 2010, All Rights Reserved houyuanhk.com.

Please input the default Username: admin; Password: admin in the presented screen above. When you login successfully, you can get the configuration web page as below:



HOUYUAN

Home Extensions PBX Features System Setup Diagnostics Admin

System Home Uptime : 20:12:50 up 1:12, load average: 0.88, 0.57, 0.31

Trunks

Status	Trunk	Type	Username	Port/Hostname/IP

Extensions

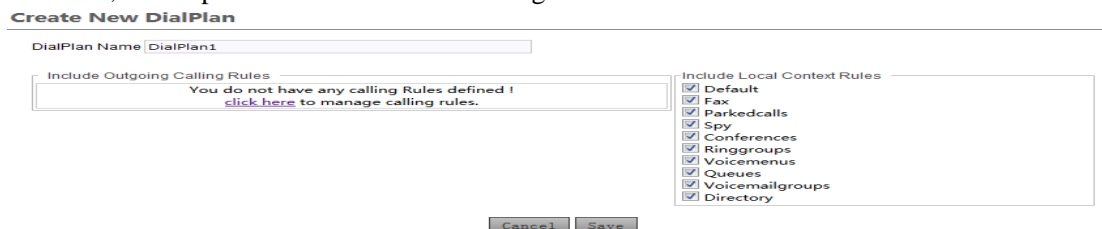
Free Busy UnAvailable Ringing

Extension	Name/Label	Status	Type
-- *No Extension assigned	Check Voicemails		VoiceMailMain
-- *No Extension assigned	Dial by Names		Directory

4.1.2 Add up Users from Web Page of IP0x

1) Add up a DialPlan

Before you add up user, you have to add up a DialPlan, please click on **Dial Plans**→**New DialPlan**, I add up a DialPlan like the following:



Create New DialPlan

DialPlan Name:

Include Outgoing Calling Rules
You do not have any calling Rules defined !
[click here](#) to manage calling rules.

Include Local Context Rules

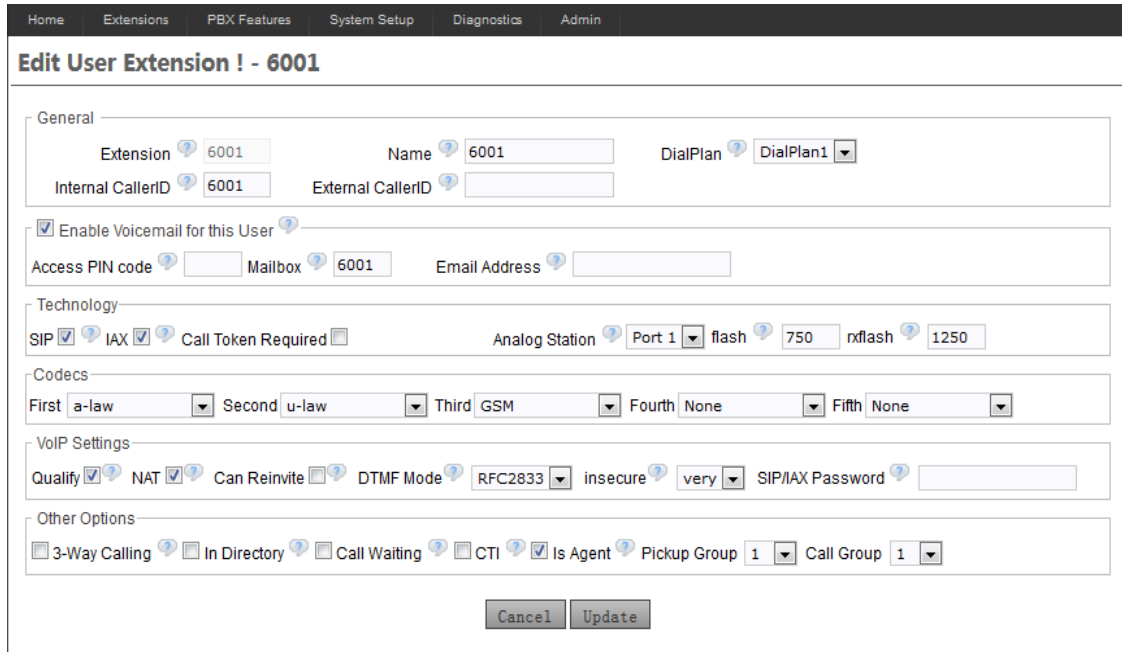
- ☒ Default
- ☒ Fax
- ☒ Parkedcalls
- ☒ Spy
- ☒ Conferences
- ☒ Ringgroups
- ☒ Voicemenus
- ☒ Queues
- ☒ Voicemailgroups
- ☒ Directory

Cancel Save

After configuring, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

2) Add up SIP user 6001

After logging into the web page of IP0x, please click on **Exten→ Create New User**, I configure user 6001 like the following:



At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3) Add up an Analog user 6005

After logging into the web page of IP0x, please click on **Exten→ Create New User**, I add a user 6005 like the following:

Home Extensions PBX Features System Setup Diagnostics Admin

Create New User

General

Extension Name DialPlan

Internal CallerID External CallerID

☒ Enable Voicemail for this User

Access PIN code Mailbox Email Address

Technology

SIP ☐ IAX ☐ Call Token Required ☐ Analog Station flash rxfash

Codecs

First Second Third Fourth Fifth

VoIP Settings

Qualify ☒ NAT ☒ Can Reinvite ☐ DTMF Mode insecure SIP/IAX Password

Other Options

☐ 3-Way Calling ☐ In Directory ☐ Call Waiting ☐ CTI ☐ Is Agent ☒ Pickup Group Call Group

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

Please pay attention to the **Technology** component, there is an **Analog Station** drop-down list, I choose port 1 in which port the analog phone plugs.

4.1.3 Register a SIP user 6001 in HY610

After logging into the web page of IP Phone HY610, please select VOIP option, I register the 6001 as the following illustration:

IP Phone

Current Status Network **VOIP** Advanced Dial-peer Config Manage Update System Manage

Public SIP Configuration

Basic Setting			
Register status	Registered	Proxy Server Address	<input type="text"/>
Server Address	192.168.1.10	Proxy Server Port	<input type="text"/>
Server Port	5060	Proxy Username	<input type="text"/>
Account Name	6001	Proxy Password	<input type="text"/>
Password	****	Domain Realm	<input type="text"/>
Phone Number	6001	Enable Register	<input checked="" type="checkbox"/>
Display Name	6001		

After configuring, please click on the **APPLY** button.

Now you can call each other directly between user 6001 and 6005.

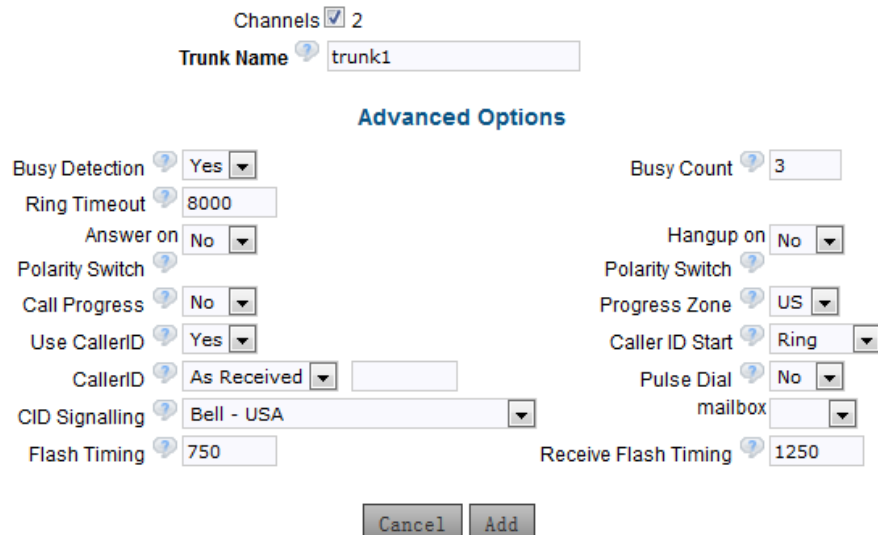
4.2 How to Make a Call to Outside through PSTN

In order to dial out to PSTN with IP0x, you need an analog trunk, an outgoing calling rule, a dial plan and a user. Here I will give the simple configuration steps which show how to make a call to outside, for detail configuration, you can refer to chapter 3.

4.2.1 Create an Analog Trunk

After logging into the web page of IP0x, please click on **Trunks**→ **Analog Trunks**, I configure an analog trunk like the following:

New Analog Trunk



At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

4.2.2 Create an Outgoing Calling Rule

After logging into the web page of IP02, please click on **Outgoing Calling Rules**→ **New Calling Rule**, I configure an outgoing calling rule like the following:

New CallingRule

Calling Rule Name

Pattern

☐ Send to Local Destination

Destination

Send this call through trunk

Use Trunk ☐ Record Calls

Strip digits from front

and Prepend these digits before dialing

☐ Use FailOver Trunk

fail over Trunk

Strip digits from front

and Prepend these digits before dialing

At last, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

4.2.3 Create a Dial Plan

After logging into the web page of IP0x, please click on **Dial Plans**→ **New DialPlan**, I configure a dial plan like the following:

Create New DialPlan

DialPlan Name

Include Outgoing Calling Rules

☒ Outgoing1

Include Local Context Rules

- ☒ Default
- ☒ Fax
- ☒ Parkedcalls
- ☒ Spy
- ☒ Conferences
- ☒ Ringgroups
- ☒ Voicemenus
- ☒ Queues
- ☒ Voicemailgroups
- ☒ Directory

At last, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

4.2.4 Create a User

I will use the user 6001 I created before, here I need to reselect a dial plan for 6001, here I need to use DialPlan2, so I select DialPlan2 in the DialPlan drop-down list.

Now I can call out with prefix 2, if the caller number is 10086, I will dial 210086.

4.3 How to Get an Incoming Call from outside

In order to get an incoming call from outside with IP0x, you need an analog trunk, an incoming calling rule, a destination (here I use IVR). Here I will give the simple configuration steps which show how to get an incoming call from outside, for detail configuration, you can refer to chapter 3.

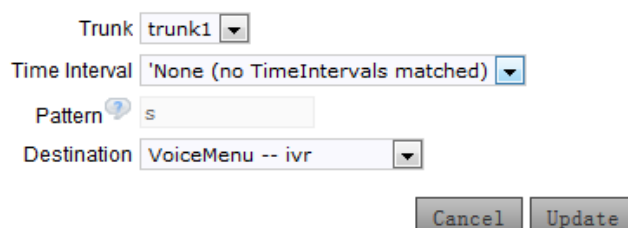
4.3.1 Create an Analog Trunk

I use the trunk1 I created in 4.2.1

4.3.2 Create an Incoming Calling Rule

After logging into the web page of IP0x, please click on **Incoming Calling Rules**→ **New Incoming Rule**, I configure an incoming calling rule like the following:

New Incoming Rule



At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

4.3.3 Create a Voice Menu

After logging into the web page of IP0x, please click on **Voice Menus**→ **Create New VoiceMenu**, I create a voice menu like the following:

Create New VoiceMenu

General Key Press Events

Name

Extension

☒ Allow Dialing Other Extensions

Actions

Answer the call

Play 1-for-am-2-for-pm & Donot Listen for KeyPress events

Goto User 6001

↓ ↑ ×

↓ ↑ ×

↓ ↑ ×

Add New Action

Key Press Events

General Key Press Events

Key	Action	Key	Action
0	--	8	--
1	Goto User 6001	9	--
2	Goto User 6005	#	--
3	--	*	--

When the call comes from port 2, the system will play a record sound file, if the caller presses 1, user 6001 will ring, if the caller presses 2, user 6005 will ring. If the caller does not press any key, the call will go to 6001.

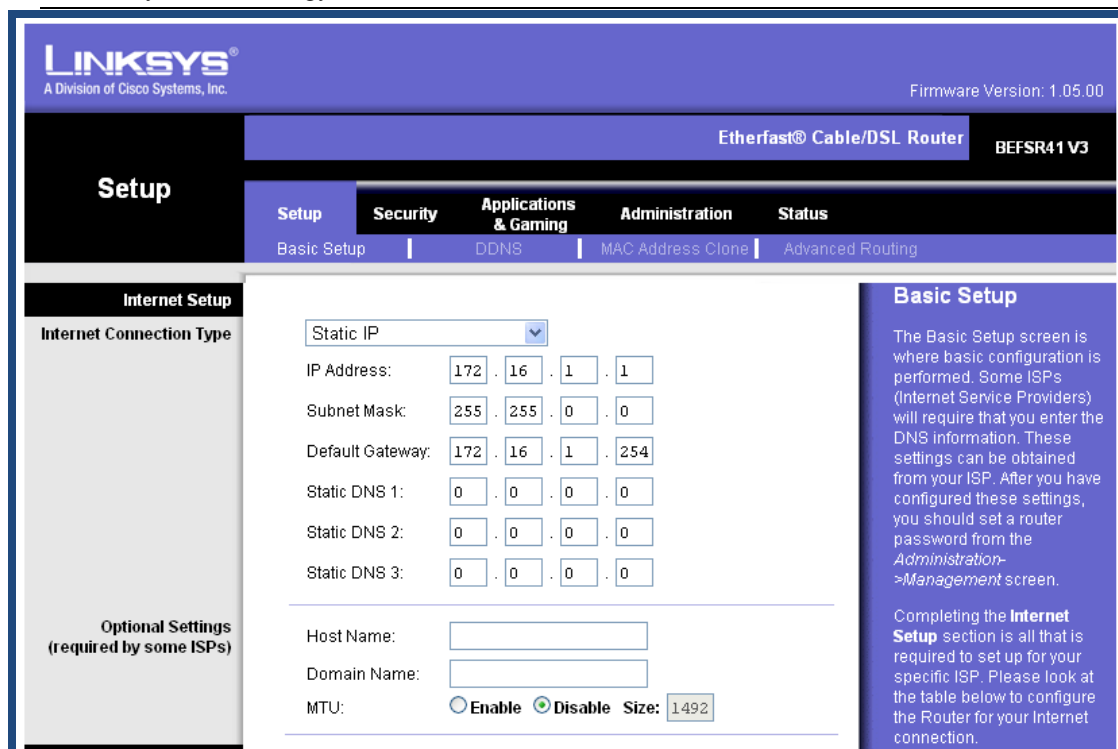
You can also configure IP0x to let 6030 call outside and get incoming call by IP0x, the steps are the same as IP0x, you can refer to configuration of IP0x.

4.4 How to Call Each Other Directly from Different Network Segment.

Take the user 6020, 6005 and 6001 for example, I will configure router, users and IP02, then the three users can call each other directly.

1) Set up router

From the web page of your router, please configure the IP address, subnet mask and default gateway of WAN port, I configured a static IP Address 172.16.1.1; Subnet Mask: 255.255.0.0; Default Gateway: 172.16.1.254. You can refer to the following:



LINKSYS®
A Division of Cisco Systems, Inc.

Firmware Version: 1.05.00

Etherfast® Cable/DSL Router **BEFSR41 V3**

Setup

Setup | Security | Applications & Gaming | Administration | Status

Basic Setup | DDNS | MAC Address Clone | Advanced Routing

Internet Setup

Internet Connection Type: Static IP

IP Address: 172 . 16 . 1 . 1

Subnet Mask: 255 . 255 . 0 . 0

Default Gateway: 172 . 16 . 1 . 254

Static DNS 1: 0 . 0 . 0 . 0

Static DNS 2: 0 . 0 . 0 . 0

Static DNS 3: 0 . 0 . 0 . 0

Optional Settings (required by some ISPs)

Host Name:

Domain Name:

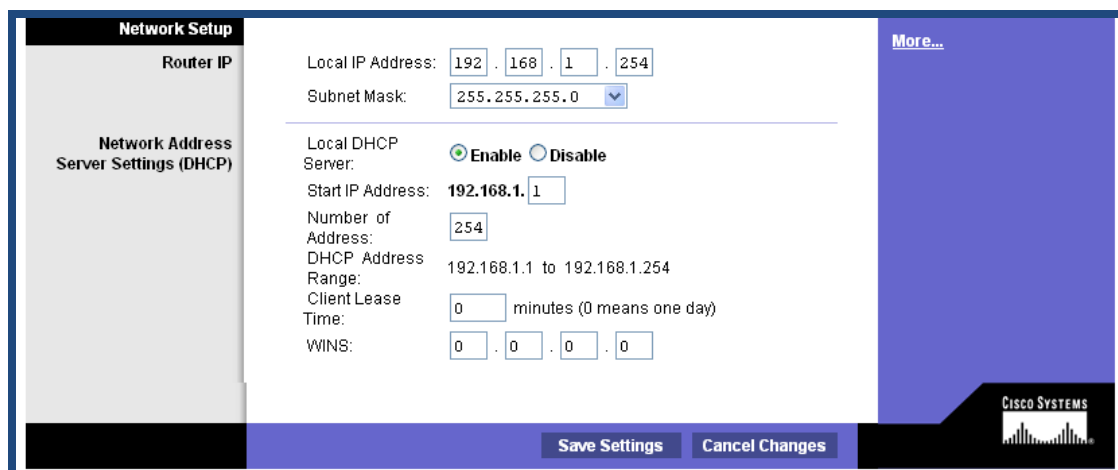
MTU: ☐ Enable ☒ Disable Size: 1492

Basic Setup

The Basic Setup screen is where basic configuration is performed. Some ISPs (Internet Service Providers) will require that you enter the DNS information. These settings can be obtained from your ISP. After you have configured these settings, you should set a router password from the Administration->Management screen.

Completing the **Internet Setup** section is all that is required to set up for your specific ISP. Please look at the table below to configure the Router for your Internet connection.

From the web page of your router, please configure the IP address, subnet mask and DHCP, I configure them like the following:



Network Setup

Router IP

Local IP Address: 192 . 168 . 1 . 254

Subnet Mask: 255.255.255.0

Local DHCP Server: ☒ Enable ☐ Disable

Start IP Address: 192.168.1.1

Number of Address: 254

DHCP Address Range: 192.168.1.1 to 192.168.1.254

Client Lease Time: 0 minutes (0 means one day)

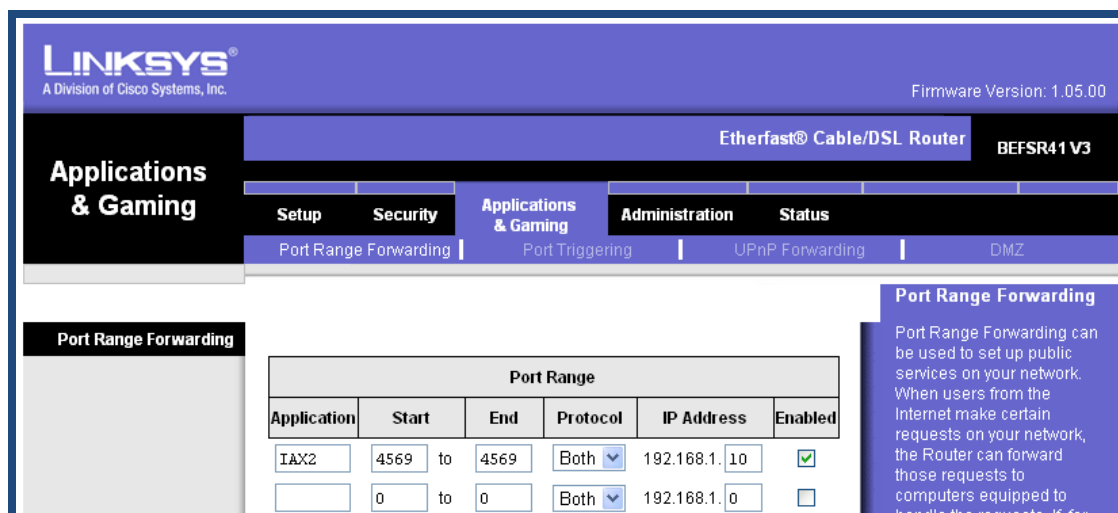
WINS: 0 . 0 . 0 . 0

More...

Save Settings Cancel Changes

CISCO SYSTEMS

From the webpage of your router, please configure port range forwarding like the following:



LINKSYS®
A Division of Cisco Systems, Inc.

Firmware Version: 1.05.00

Etherfast® Cable/DSL Router **BEFSR41 V3**

Applications & Gaming

Setup Security **Applications & Gaming** Administration Status

Port Range Forwarding Port Triggering UPnP Forwarding DMZ

Port Range Forwarding

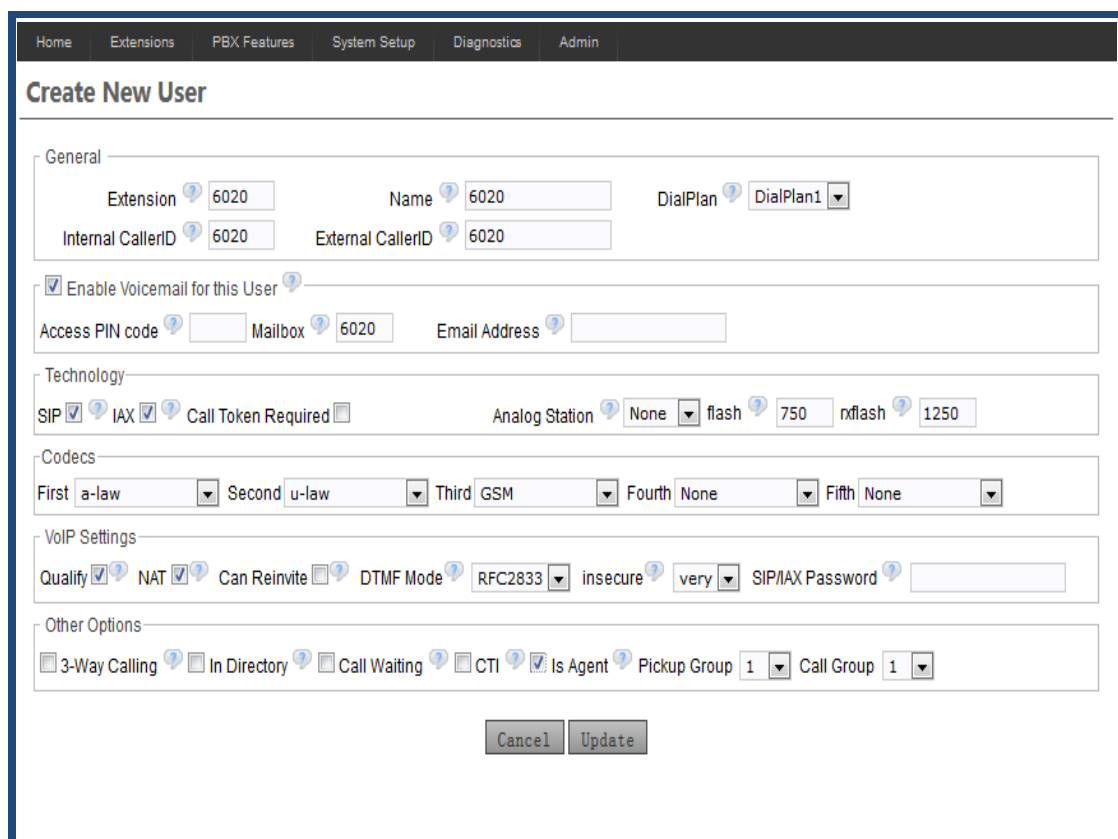
Port Range Forwarding can be used to set up public services on your network. When users from the Internet make certain requests on your network, the Router can forward those requests to computers equipped to handle the requests. If you

Port Range					
Application	Start	End	Protocol	IP Address	Enabled
IAX2	4569	to 4569	Both	192.168.1.10	<input checked="" type="checkbox"/>
	0	to 0	Both	192.168.1.0	<input type="checkbox"/>

The user 6020 uses IAX2, the port number is 4569, 192.168.1.10 is the IP address of IP0x.

2) Add an IAX user 6020 in IP0x

After logging into the web page of IP0x, please click on **Users** → **Create New User**, I configure 6020 like the following:



Create New User

Home Extensions PBX Features System Setup Diagnostics Admin

General

Extension 6020 Name 6020 DialPlan DialPlan1

Internal CallerID 6020 External CallerID 6020

☒ Enable Voicemail for this User

Access PIN code Mailbox 6020 Email Address

Technology

SIP ☒ IAX ☒ Call Token Required ☐ Analog Station None flash 750 nflash 1250

Codecs

First a-law Second u-law Third GSM Fourth None Fifth None

VoIP Settings

Quality ☒ NAT ☒ Can Reinvite ☐ DTMF Mode RFC2833 insecure very SIP/IAX Password

Other Options

☐ 3-Way Calling ☐ In Directory ☐ Call Waiting ☐ CTI ☒ Is Agent Pickup Group 1 Call Group 1

Cancel Update

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

3) Set up HY620 and register an IAX2 user 6020

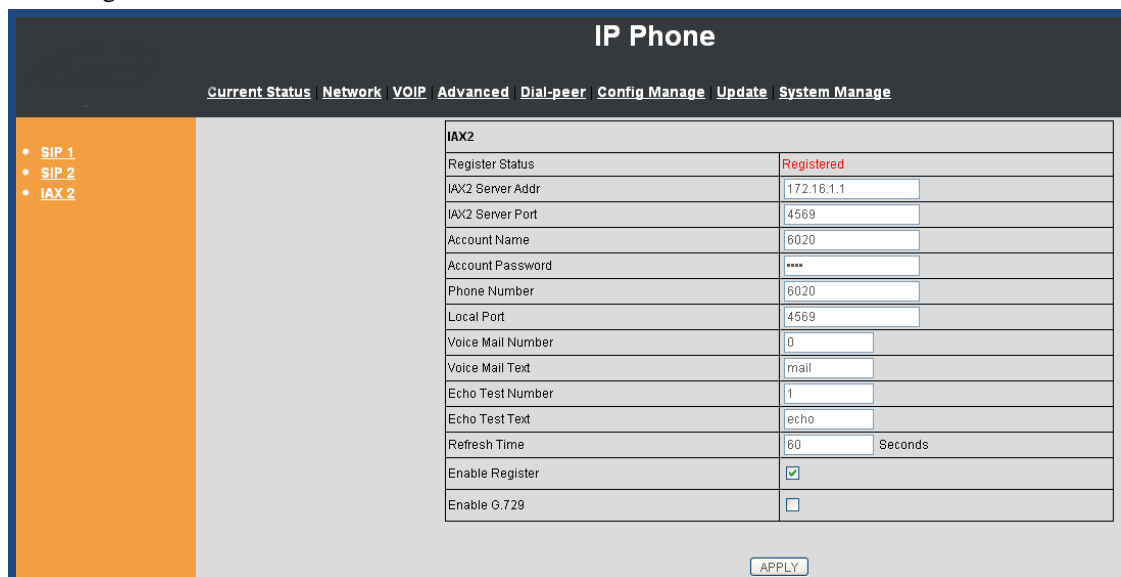
After logging into web page of IP Phone HY-620, please select **Network** option to enter the screen of configuring IP Address. I set up a static IP Address: 172.16.1.2; Netmask: 255.255.0.0; Gateway: 172.16.1.254. After finishing the configuration, please click on the **Apply** button. You

can refer to the following screen:



WAN Status	
Active IP	172.16.1.2
Current Netmask	255.255.0.0
Current Gateway	172.16.1.254
MAC Address	00:09:45:56:fd:ce
Get MAC Time	20090915
WAN Setting	
Static <input checked="" type="radio"/>	DHCP <input type="radio"/> PPPOE <input type="radio"/>
Auto DNS	<input checked="" type="checkbox"/>
Static IP Address	172.16.1.2
Netmask	255.255.0.0
Gateway	172.16.1.254
DNS Domain	
Primary DNS	202.96.134.133
Alter DNS	202.96.128.68

Please select the **VOIP** option, then select the **IAX2** option, I register the IAX2 user 6020 as the following illustration:



IAX2	
Register Status	Registered
IAX2 Server Addr	172.16.1.1
IAX2 Server Port	4569
Account Name	6020
Account Password	****
Phone Number	6020
Local Port	4569
Voice Mail Number	0
Voice Mail Text	mail
Echo Test Number	1
Echo Test Text	echo
Refresh Time	60 Seconds
Enable Register	<input checked="" type="checkbox"/>
Enable G.729	<input type="checkbox"/>

After configuring, please click on the **APPLY** button.

Attention: here you must register IAX2 user instead of SIP user, because the user 6020 is not in the same network segment as IP0x. If you use SIP user, you can not get sound when the communication is established.

Now you can call each other among 6020,6001 and 6005 directly.

4.5 How to Call through VoIP Trunk

4.5.1 Call from IP02 to IP08

In order to call from IP02 to IP08, I will create a SIP user in IP08 for the SIP trunk in IP02, create

a SIP trunk, an outgoing call rule and a dial plan in IP02.

- 1) Add an SIP user 6035(it will be used as SIP trunk in IP02) in IP08, after logging into the web page of IP08, please click on **Users→ Create New User**, I add the user 6035 like the following:

Create New User

General

Extension ? 6035

Name ? 6035

DialPlan ? DialPlan1

Internal CallerID ? 6035

External CallerID ? 6035

☒ Enable Voicemail for this User ?

Access PIN code ?

Mailbox ? 6000

Email Address ?

Technology

☒ SIP ? ☒ IAX ? ☐ Call Token Required

Analog Station ? None flash ? 750 rflash ? 1250

Codecs

First a-law

Second u-law

Third GSM

Fourth None

Fifth None

VoIP Settings

Quality ? ☒ NAT ? ☒ Can Reinvite ? ☐ DTMF Mode ? RFC2833 insecure ? very

SIP/IAX Password ? 6035

Other Options

☐ 3-Way Calling ? ☐ In Directory ? ☐ Call Waiting ? ☐ CTI ? ☒ Is Agent ?

Pickup Group 1

Call Group 1

Cancel

Update

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

Add a SIP user 6030 in IP08 for HY620, the way is the same as adding 6035.

- 2) Add a VoIP trunk in IP02, after logging into the webpage of IP02, please click on **Trunks→VOIP Trunks→New SIP/IAX Trunk**, I configure a SIP trunk1 like the following:

Create New SIP/IAX trunk

Type

Use routing context ☐

Provider Name

Hostname :

Username

Authuser

Fromuser

Fromdomain

Password

Contact

Quality ☒

Insecure Type

After configuring, please click on **Add** button, and click on **Apply Changes** button in up right corner of the main page.

- 3) Create an outgoing calling rule in IP02, after logging into the webpage of IP02, please click on **Outgoing Calling Rules**→**New Calling Rule**, I configure an outgoing2 rule like the following:

New CallingRule

Calling Rule Name

Pattern

☐ Send to Local Destination

Destination

Send this call through trunk

Use Trunk ☐ Record Calls

Strip digits from front

and Prepend these digits before dialing

☐ Use FailOver Trunk

fail over Trunk

Strip digits from front

and Prepend these digits before dialing

After configuring, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

- 4) Create a dial plan in IP02, after logging into the webpage of IP02, please click on **Dial Plans**→**New DialPlan**, I configure a dialplan2 like the following:

Edit DialPlan

DialPlan Name

Include Outgoing Calling Rules

- ☒ Outgoing1
- ☒ Outgoing2

Include Local Context Rules

- ☒ Default
- ☒ Fax
- ☒ Parkedcalls
- ☒ Spy
- ☒ Conferences
- ☒ Ringgroups
- ☒ Voicemenus
- ☒ Queues
- ☒ Voicemailgroups
- ☒ Directory

After configuring, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

In configuration screens of 6001 and 6005, please select dialplan1 in the **DialPlan** drop-down list
Now you can call from 6001 and 6005 to 6030 by dialing 96030








































4.5.2 Call from IP08 to IP02

In order to call from IP08 to IP02, I will create a SIP user in IP02 for the SIP trunk in IP08, create a SIP trunk, an outgoing call rule and a dial plan in IP08.

- 1) Add a user 6008 in IP02

Add a SIP user: 6008, after logging into the web page of IP02, please click on **Users**→**Create New User**, I add a user 6008 like the following:

Create New User






General		
Extension 	6008	Name 
Internal CallerID 	6008	External CallerID 
		DialPlan 
		DialPlan1 
<input type="checkbox"/> Enable Voicemail for this User 		
Access PIN code 		Mailbox 
		6000
		Email Address 
Technology		
SIP 	<input checked="" type="checkbox"/>	IAX 
	<input checked="" type="checkbox"/>	Call Token Required 
		Analog Station 
		None 
		flash 
		750
		rxflash 
		1250
Codecs		
First	A-law 	Second
	U-law 	Third
	GSM 	Fourth
	None 	Fifth
	None 	
VoIP Settings		
Quality 	<input checked="" type="checkbox"/>	NAT 
	<input checked="" type="checkbox"/>	Can Reinvite 
	<input type="checkbox"/>	DTMF Mode 
		RFC2833 
		insecure 
		no 
		SIP/IAX Password 
		6008
Other Options		
<input type="checkbox"/> 3-Way Calling 	<input type="checkbox"/> In Directory 	<input type="checkbox"/> Call Waiting 
<input type="checkbox"/> CTI 	<input checked="" type="checkbox"/> Is Agent 	Pickup Group 
		1 
		Call Group 
		1 
<div> <div>Cancel</div> <div>Update</div> </div>		

At last, please click on **Update** button, and click on **Apply Changes** button in up right corner of the main page.

2) Create a SIP trunk in IP08

Add a VoIP trunk in IP08, after logging into the webpage of IP08, please click on **Trunks**→**VOIP Trunks**→**New SIP/IAX Trunk**, I configure a SIP trunk like the following:

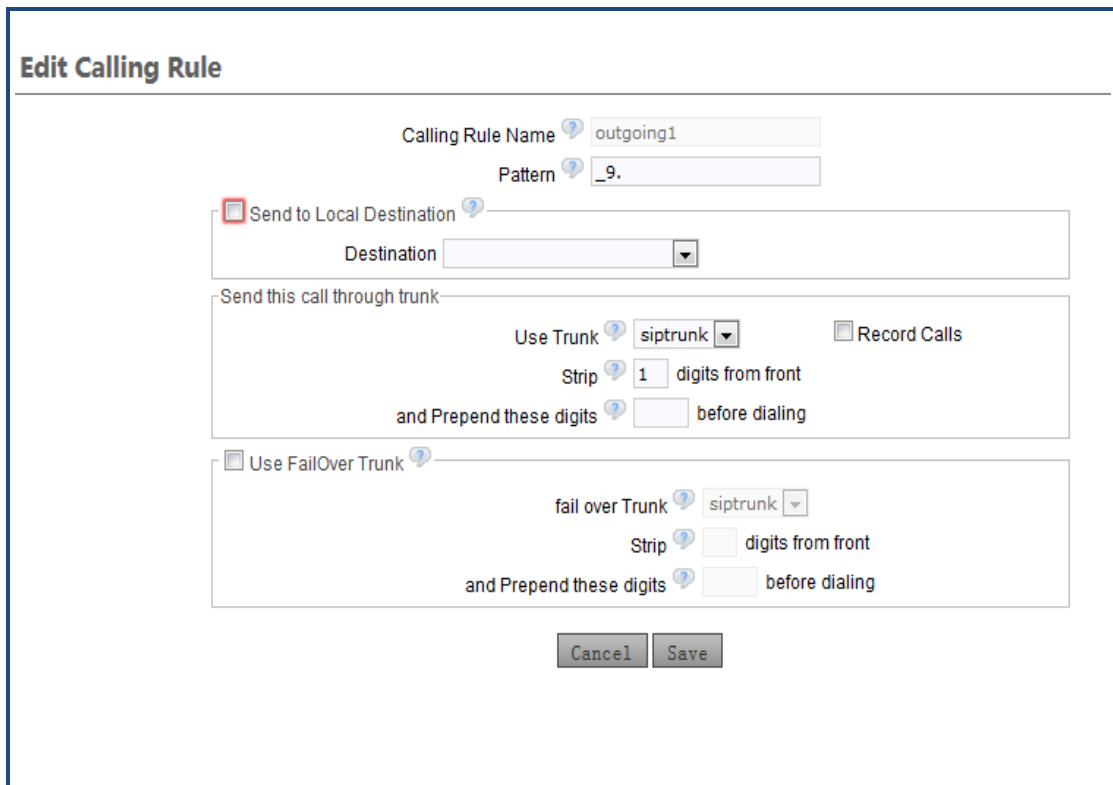
Create New SIP/IAX trunk

Type	SIP 
Use routing context 	<input type="checkbox"/>
Provider Name 	siptrunk
Hostname	192.168.1.10 : 5060
Username	6008
Authuser	
Fromuser	
Fromdomain	
Password	6008
Contact	
Quality 	<input checked="" type="checkbox"/> 2
Insecure Type	very 
<div> <div>Cancel</div> <div>Add</div> </div>	

After configuring, please click on **Add** button, and click on **Apply Changes** button in up right corner of the main page.

3) Create an outgoing calling rule in IP08

After logging into the webpage of IP08, please click on **Outgoing Calling Rules**→**New Calling Rule**, I configure an outgoing1 rule like the following:



After configuring, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

4) Create a dial plan in IP08

After logging into the webpage of IP08, please click on **Dial Plans**→**New DialPlan**, I configure a dialplan1 like the following:



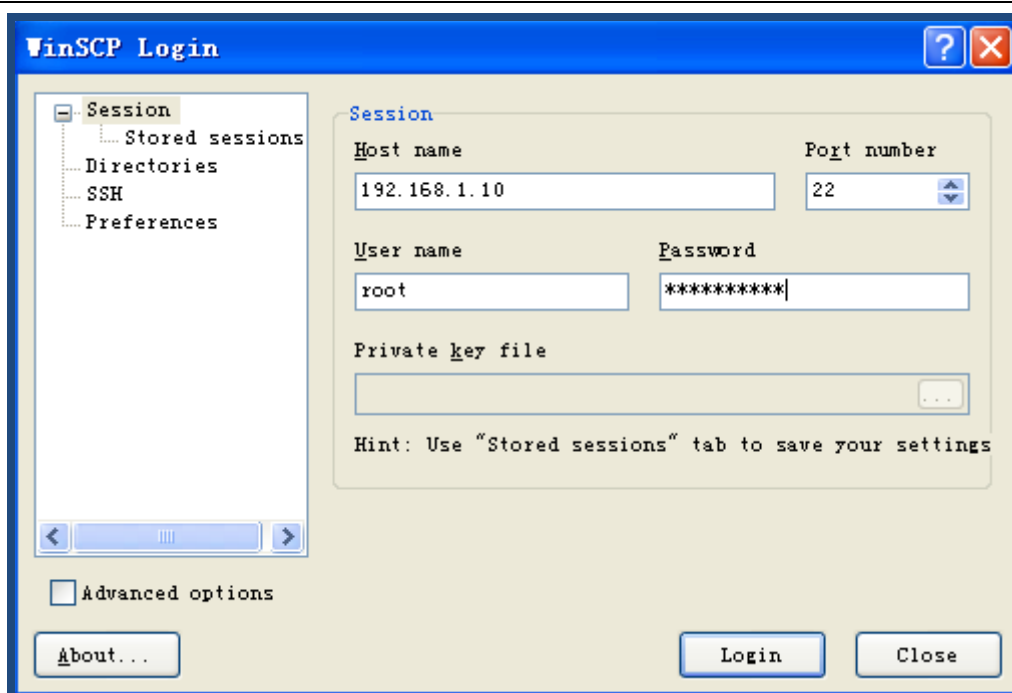
After configuring, please click on **Save** button, and click on **Apply Changes** button in up right corner of the main page.

In configuration screens of 6030, please select dialplan1 in the **DialPlan** drop-down list.

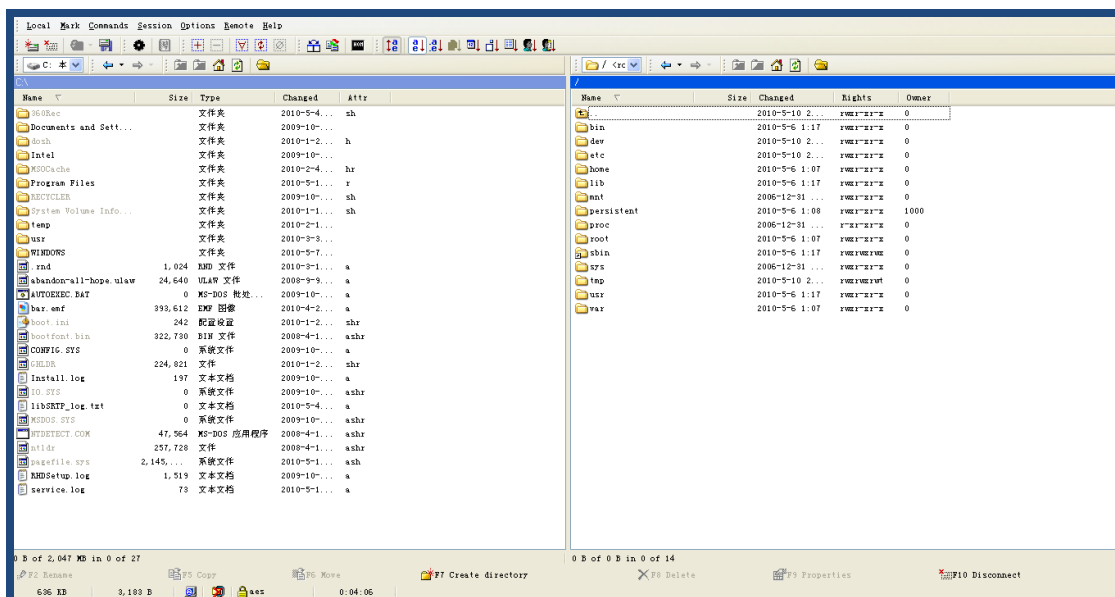
Now you can call from 6030 to 6001 and 6005 by dialing with prefix 9.

4.6 How to Transfer Files between Windows PC and IP0x

Using WinSCP software, it is the most convenient way to transfer files between windows PC and IP0x. Open your WinSCP software, enter the IP Address, username, password of IP0x like the following screen:

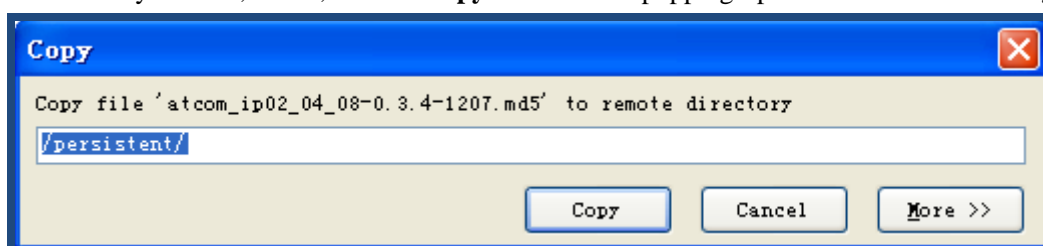


At last, click on **Login** button, then you can get the following screen:



The left part of the screen displays directories and files of your windows PC, the right part of the screen displays directories of IP0x.

If you want to transfer a file from windows PC to IP0x, you just need to choose the file and drag it to the directory of IP0x, at last, click on **copy** button in the popping-up screen like the following:



Chapter 5 Reference

<http://www.houyuanhk.com>

<http://www.globalsources.com/houyuanhk.co>

